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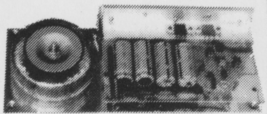
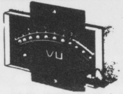


MF100



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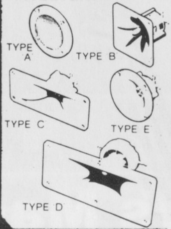

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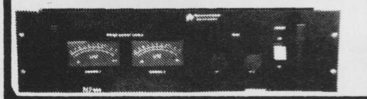

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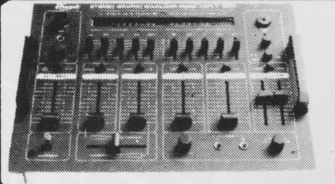

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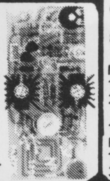

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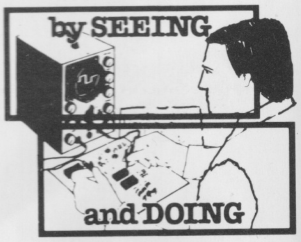

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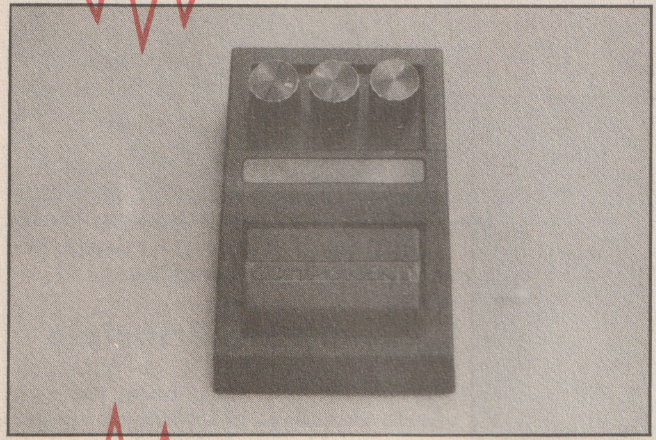

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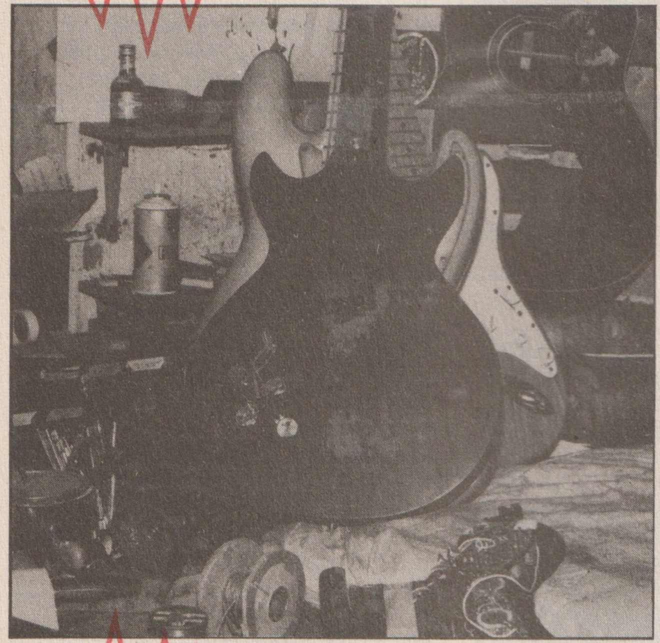

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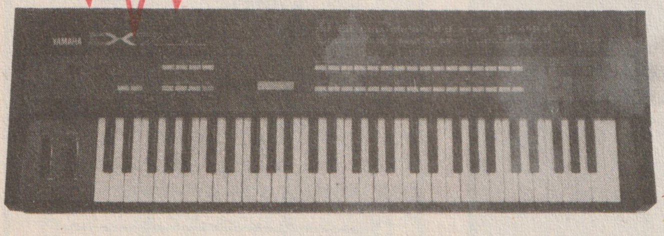

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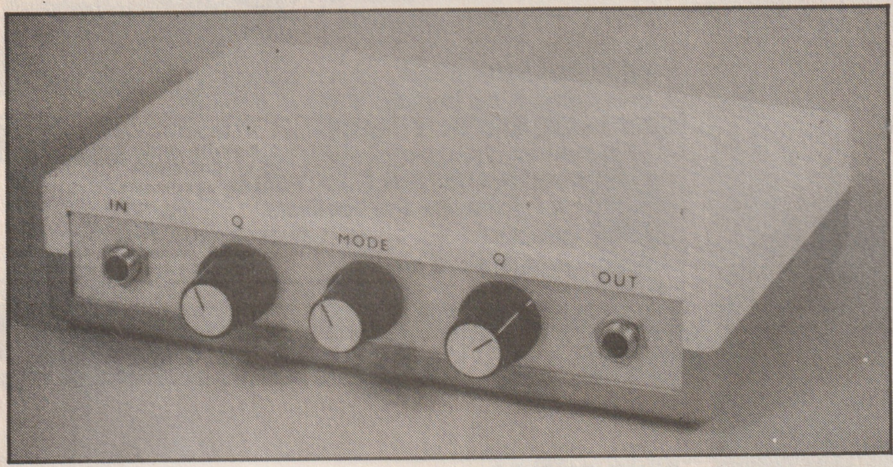
COMPUTER CONTROLLED FILTER



There are various ways of controlling the filter frequency of a musical effect. This one uses a Commodore 64.

R. A. Penfold

NORMAL FILTER type musical effects units (phase and waa-waa units for example) use a low frequency oscillator to provide automatic sweeping of the filter frequency. Another approach is to use the envelope of the input signal or the control voltage from an envelope shaper (as in a synthesiser) to provide the sweeping. A third method is to simply use a foot pedal to control the effect. This novel effects unit uses yet another approach, and this is to have a computer to control the filter frequency. Although only a relatively crude method of control is possible, this enables a useful range of effects to be obtained, and the versatility is increased by having two filters. Notch, bandpass, and lowpass filtering are available, with each filter having a Q or resonance control. This enables simple phasing plus two forms of waa-waa effect to be produced.

While perhaps not being ideal as a guitar effects unit, this project is useful to anyone interested in sound synthesis, particularly in stereo set-ups. For stereo operation the main way of using the unit is to feed one signal to both filter inputs, and then take a pseudo stereo output from the two filter outputs. This would be pointless if the filters were to be swept up and down in frequency in unison, but some interesting effects can be obtained if (for example) the filters are operated in antiphase, with the frequency of one being swept downwards while the other is swept upwards. The effective sweep waveform can be altered under software control, and can be triangular, sawtooth (of either type), squarewave, or even a random modulation of the filters can be used if desired. The two filters do not need to be set for the same type of filtering, and interesting results can be produced using (say) bandpass filtering in one channel and notch filtering in the other. Each filter has a separate input, and the unit can be fed from a stereo source if desired. Obviously there are numerous ways in which the unit can be used, and it is ideal for the more adventurous electronic music enthusiast.

The unit is primarily intended for use with the Commodore 64 computer, but it will work in a more basic manner with the VIC-20 or BBC model B, or any computer which has a user port provided by port B of a 6526 or 6522 device. With machine code software it should be possible to drive it from any computer that can provide one or two digital outputs, but this goes beyond the scope of the present article.

Operating Principle

One approach to a computer controlled filter would be to have an ordinary voltage controlled filter driven from the computer via a digital to analogue converter. A more simple alternative, and the one adopted for this design, is to use a switched capacitor filter with the computer providing the clock signal. Switched capacitor filters may be unfamiliar to many readers, and while not being a particularly new concept, have received relatively little use.

An ordinary single stage lowpass filter just consists of a resistor and a capacitor connected as shown in **Figure 1(a)**. Ra and Ca form a potential divider circuit, but the degree of attenuation through the circuit is not fixed, and depends on the impedance of Ca. This decreases as the input frequency is increased, giving losses that rise as the input frequency is increased. The losses are minimal at first, but once the 6dB point has been reached (the output potential is half the input voltage) the attenuation rate is 6dB per octave (doubling the input frequency halves the circuit gain).

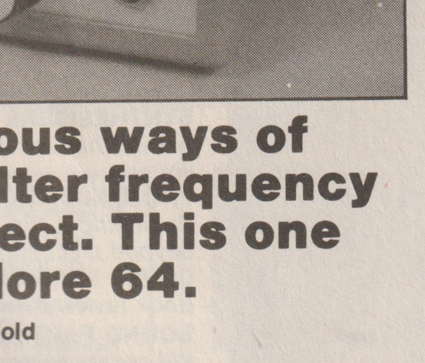


Figure 1a. A simple low pass filter circuit.

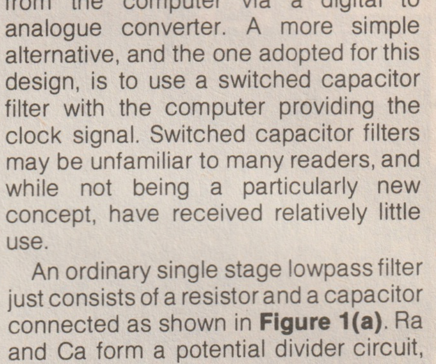
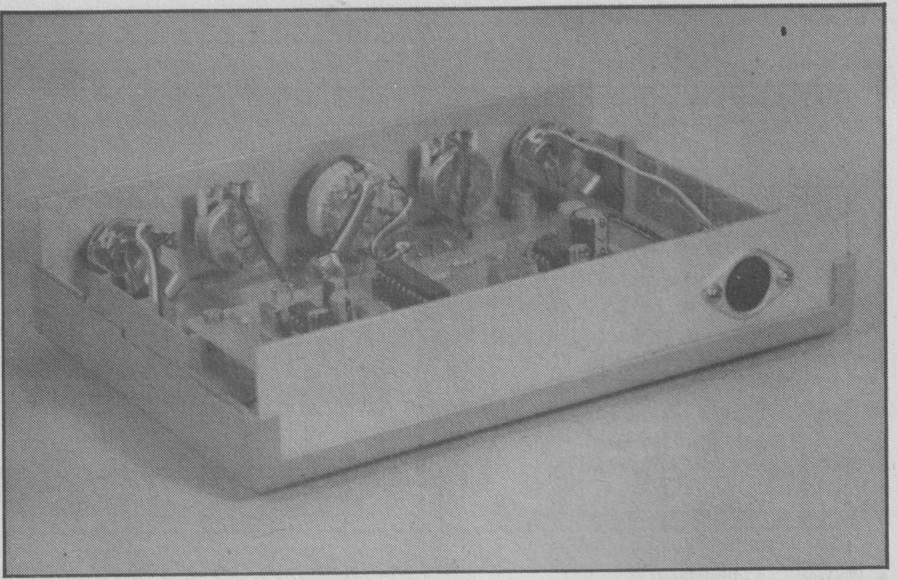


Figure 1b. The circuit for a switched capacitor filter.



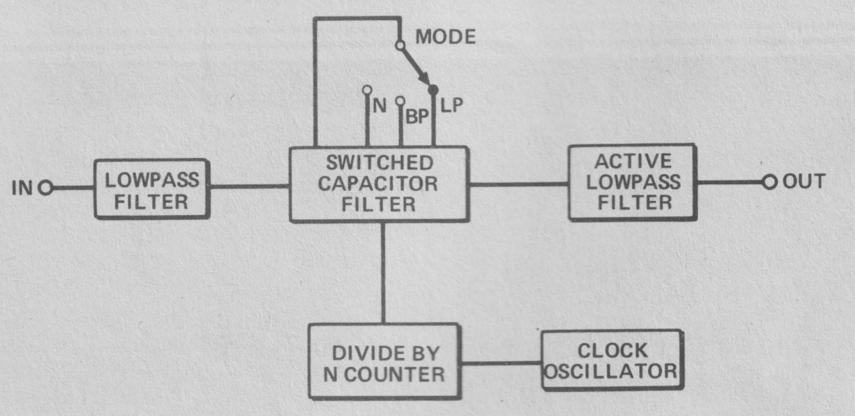


Figure 2. The block diagram for one channel of the filter: the other channel is identical.

A switched capacitor filter operates in a similar fashion, but the resistor is replaced with a low value capacitor and an electronic (SPDT) switch, as in **Figure 1(b)**. Cb and SW1 are the capacitor and the switch, while Cc is the normal filter capacitor. A clock oscillator controls the rate at which the electronic switch alternates between the position where it takes a charge from the input, and is discharged into Cc. A change in the potential at the input of the circuit is obviously transferred to the output of the circuit by the charge/discharge action of Cb and SW1, but changes in potential are not instantly transferred. The time taken for the output of the circuit to respond to changes at the input depends on the relative size of Cb and Cc, and on the rate at which SW1 is clocked. it is this last point which is of importance, since raising the clock frequency provides a faster transfer, and boots the cutoff frequency of the filter. In other words, the filter's cutoff frequency is proportional to the clock frequency.

In a practical switched capacitor filter there are usually two filters wired in series, and some other circuits are included so that the basic lowpass filter action can be modified to give any other form of filtering. The switched capacitor device used in this project is the National Semiconductors MF10CN which can operate in any filter mode, has adjustable Q, and two 12dB per octave filters. The filter frequency can be either 1/50th or 1/100th of the clock frequency but it must be the same for both filters.

Circuit Operation

The block diagram of **Figure 2** shows the make-up of one channel of the unit (the other channel is identical). A passive lowpass filter is used ahead of the switched capacitor filter, and this is needed to prevent strong signals close to the clock frequency from entering the switched capacitor filter and producing heterodyne 'whistles'. The switched-capacitor filter operates in a mode which give notch, bandpass, and lowpass filtering, and a switch is used to select the desired mode. Clock glitches only appear at the output of the filter at a relatively low level, but an active filter here gives further attenuation and a clean output signal. This stage also provides output buffering.

The clock signal is provided by the system clock of the computer, and is at a frequency of about 1MHz, but the precise frequency depends on the particular computer used (980kHz in the case of the Commodore 64). It is connected to the filter via a divide by N counter, but this is also part of the computer and not included in this project. The divider can be programmed to divide by any even integer from 2 to 131070, and a vast range of some 65535 output frequencies is therefore available. However, most of these are unusable in this case as they are within the audio range, and would produce noticeable breakthrough at the output. The available range enables some useful effects to be generated though.

The full circuit diagram of the unit appears in **Figure 3**. If we take the channel based on IC1a, R1 and C2 are the input filter, while IC2 operates as a standard third order 18dB per octave) lowpass filter at the output. SW1a selects the required type of filtering, and on the prototype this is ganged with

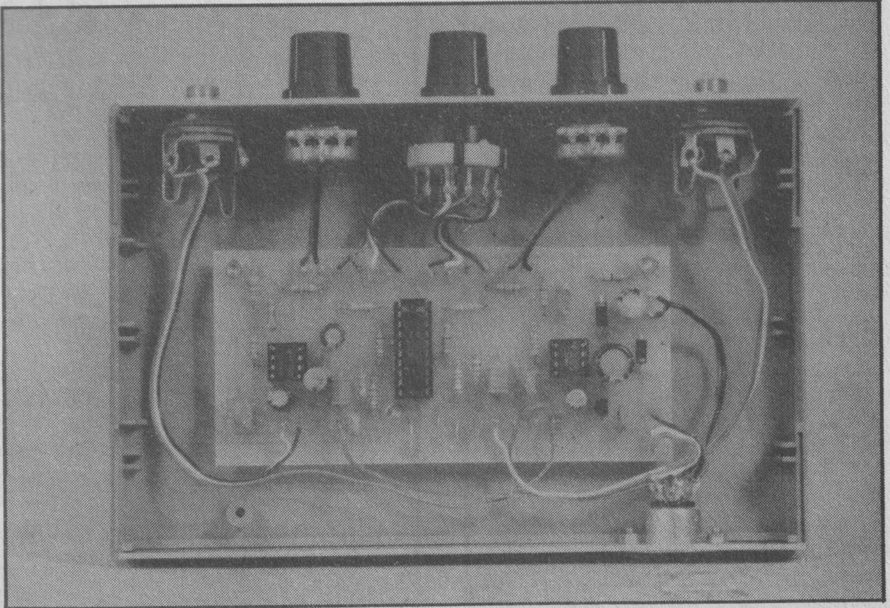
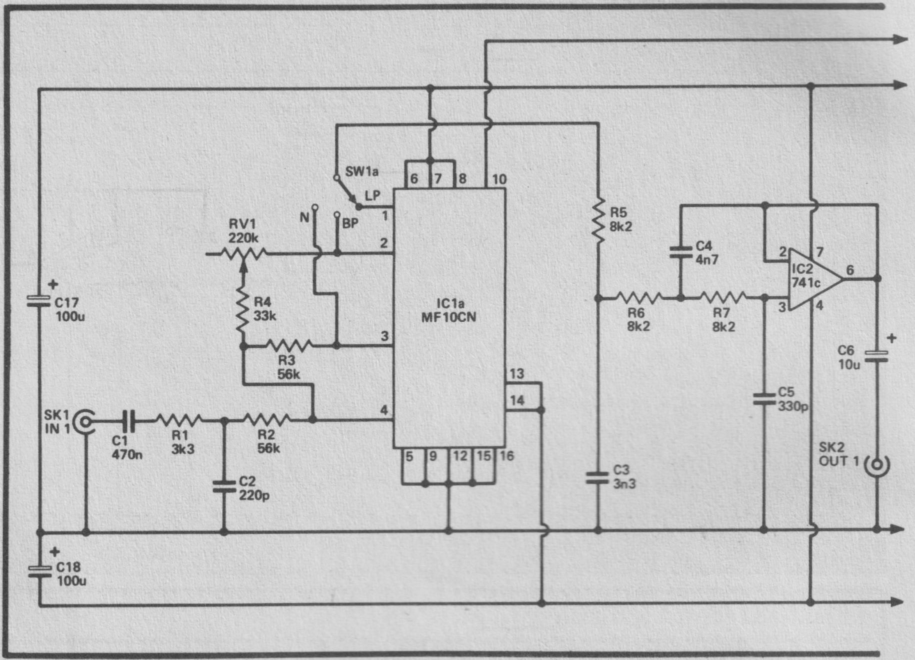


Figure 3: The circuit. The switches should be used in place of SW2 if you intend to use the two channels in different filter modes.


SW1a
LP
N
BP
RV1
220k
R4
33k
R3
56k
C17
100u
SK1
IN 1
C1
470n
R1
3k3
R2
56k
C2
220p
C18
100u
6
7
8
10
1
2
3
4
IC1a
MF10CN
13
14
5
9
12
15
16
R5
8k2
C4
4n7
R6
8k2
R7
8k2
IC2
741c
2
7
6
3
4
C6
10u
C5
330p
C3
3n3
SK2
OUT 1

Parts list

RESISTORS

(All ¼W 5%)

R1,8	3k3
R2,3,9,10	56k
R4,11	33k
R5,6,7,12,13,14	8k2
R15	22R

POTENTIOMETERS

RV1,2	220k linear

CAPACITORS

C1,7	470nF carbonate
C2,8	220pF ceramic plate
C3,9	3n3 carbonate
C4,10	4n7 carbonate
C5,11	330pF ceramic plate
C6,13	10uF 25V radial elect
C12,15	100nF ceramic
C14	470uF 16V radial elect
C15	220uF 16V radial elect
C17,18	100uF 10V radial elect

SEMICONDUCTORS

IC1	MF10CN
IC2,3	741C
IC4	79L05 -5V 100mA regulator
D1,2	1N4002

MISCELLANEOUS

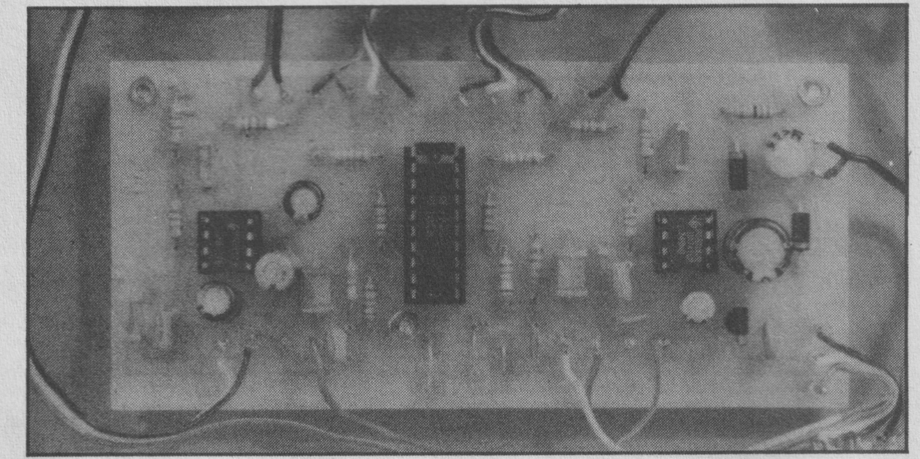
SW1	4 pole 3 way rotary
SK1,2,3,4	standard jacks or stereo jacks
SK5	5 way (180 degree) DIN socket

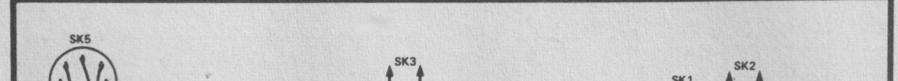
Case about 180 × 120 × 39mm; three control knobs; 5 way cable, 5 pin (180 degree) DIN plug, and 2 × 12 way 0.1 inch pitch female edge connector; 20 pin DIL IC holder; wire, solder, etc.

Shop

Most of the components, including the MF10CN, are available from Maplin (0702 552911) and other general component suppliers. The components should cost around £14 exclusive of PCB and case.

The PCB will become available from our PCB service.




SK5
PB7
+5V
PB6
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9V AC
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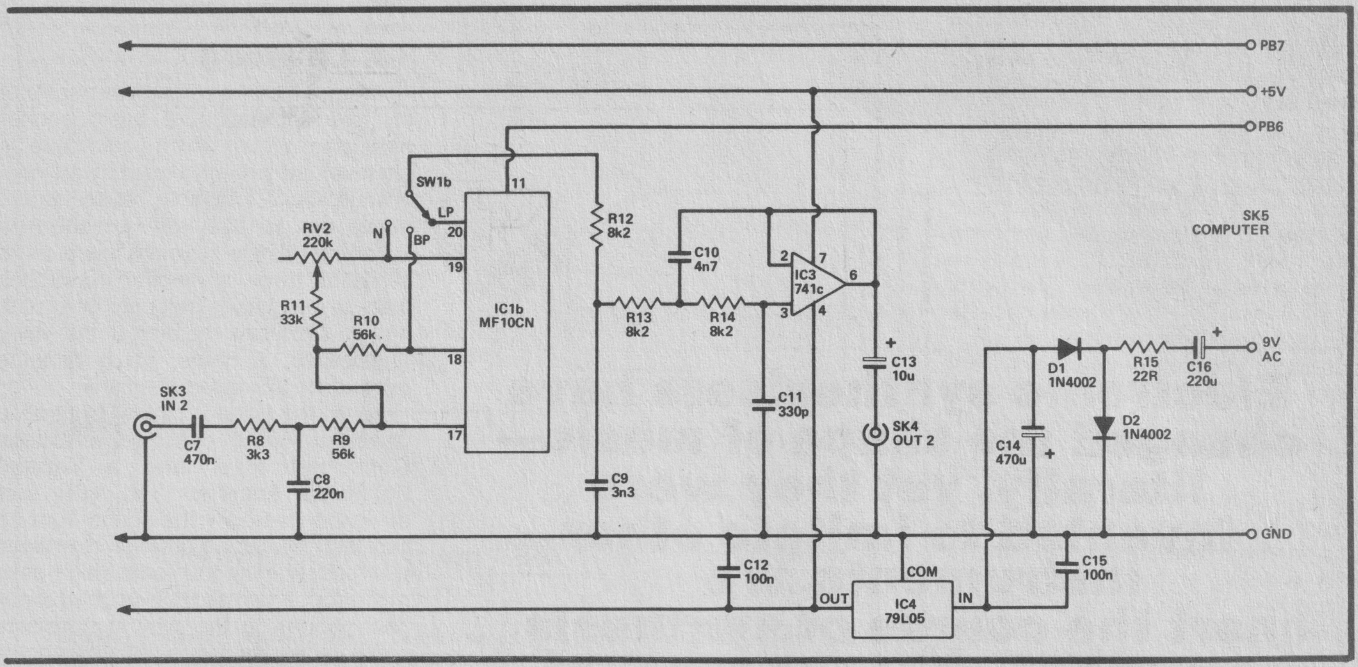
SK3
(E)
SK1
(E)
SK2
(E)
C15
IC4
C13
C12
C8
C11
C7
C9
R8
R9
R10
IC1
R2
R1
C2
C6
C3
C5
C14
IC3
IC2
D2
D1
C16
R7
R3
C1
R11
R4
R12
R6
C10
C4
R5
R15
A
B
C
D
RV2
RV1
B
C
D
A
SW1

Figure 4. The component layout: straightforward, but use an IC socket for IC1.



SW1b in the other channel. However, separate switches should be used here if you wish to use the two channels in different filter modes. RV1 is the 'Q' control. The filters are connected to operate with a 100 to 1 clock/cutoff frequency ratio, which gives an effective sweep range of about 150Hz to 5kHz.

The circuit requires dual balanced 5 volt supplies. The +5 volt supply is available from the Commodore 64's user port, but the -5 volt supply has to be derived from one of the 9 volt AC outputs using a simple rectifier, smoothing, and regulator circuit based on IC4.

Construction

Details of the printed circuit board and wiring are shown in **Figure 4**. There is nothing unusual about construction of the board, but note that IC1 is a MOS device and should be mounted in a holder. The other MOS handling precautions should also be observed. The connections to the computer are taken via a 5 way DIN socket fitted on the rear panel of the case. From here the connections are carried to the computer by way of a 5 way ribbon cable up to about a metre long and fitted with a 5 way DIN plug to connect to SK5, and a 2 by 12 way edge connector at the end which connects to the user port of the computer. **Figure 5** gives wiring details for the edge connector.

The case for the prototype is a plastic Verocase having metal front and rear panels, with approximate outside dimensions of 180 by 120 by 39 millimetres. Any case of roughly the same size should take all the parts without any difficulty though. The controls and input/output sockets are mounted on the front panel. I used a stereo jack socket at the inputs, and another at the outputs, but any type of audio connector that is convenient for your particular set-up can be used.

In Use

It is advisable to plug the unit into the computer prior to switching on the latter. The filters have a medium input impedance, low output impedance, and nominally unity voltage gain, and they should therefore fit into most set-ups with no problems. Signals of up to about 2 volts RMS can be handled.

In order to set the timer/counters of the computer's interface chip to the correct operating mode a value of 23 must be written to the two control registers. These are at addresses 56590 and 56591. The high bytes of the two timer/counters are at addresses 56581 and 56583, and in this application these are set to and simply left at zero. The filter frequencies are then controlled by the values written to the low bytes at addresses 56580 and 56582. These control the filters fed from PB6 and PB7 respectively. Values from 1 to about 25 can be used, and although this gives rather crude control of the filter frequency, it nevertheless produces some good effects. This simple program demonstrates the basic way in which the filter is used.

```
10  POKE 56590,23
20  POKE 56591,23
30  POKE 56581,0
40  POKE 56583,0
50  R=25
60  FOR L = 1 TO 25
70  POKE 56580,L:POKE 56582,R
80  R=R-1
90  NEXT
100 FOR L = 25 TO 1 STEP -1
110 POKE 56580,L:POKE 56582,R
120 R=R+1
130 NEXT
140 GOTO 60
```

This program moves the filter frequencies up and down in antiphase, using what is effectively a triangular modulation waveform. You do not really need to be a software genius in order to work out simple routines to vary the filter frequencies in various ways, and timing loops or dummy instructions can be added in order to reduce the modulation frequency. If you are familiar with 6510 machine code programming then it is possible to obtain faster operation if desired.

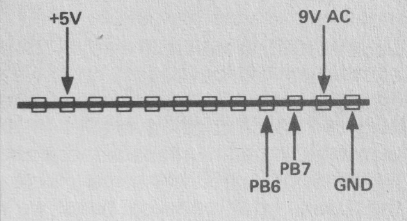


Figure 5. Wiring details for the edge connector.

With the VIC-20 computer both filters must be driven from PB7 of the user port. A value of 192 is written to the control register at address 37147 in order to set the timers to the correct mode, 0 is written to the high byte of the timer/counter at 37141, and the filter frequency is controlled by writing a value in the range 1 to about 25 or so to the low byte at 37140.

The BBC model B computer is used in the same way, but the control register, high byte, and low byte addresses are &FE6B, &FE65, and &FE64 respectively. The user port of this machine uses a different connector (a 20 way IDC type), and it does not provide a 9 volt AC output. However, a -5 volt supply is available from the power port, and this could be used to provide the negative supply with the rectifier and smoothing component being omitted.

As pointed out earlier, any computer with one or two digital outputs could probably provide suitable clock signals with the aid of a machine code program.

SYNTHESIZERS

Electronic synthesisers have changed the shape of music—literally, yet they were invented to imitate other instruments. We chart the course of synthesis from the start to the present.

Mark Jenkins

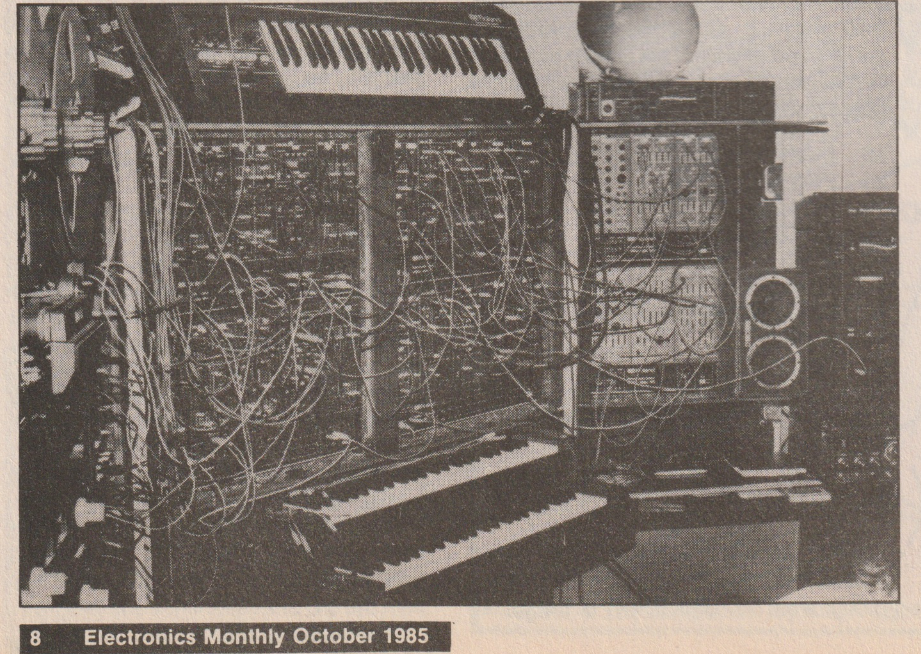
ELSEWHERE in this issue we are looking at musical effects boxes; here we're taking a look at the most sophisticated kind of "effects box" available — the electronic music synthesizer. Certainly in the early days, synthesizers weren't the sleek, wood-panelled beast we've become used to in the past few years. But what are music synthesizers, and why were they invented?

Once we've dealt with those questions, we can look in a little more detail at how synthesizers work, and think about what contributions we can make to their design and construction.

Making sounds, and latterly musical sounds, has always been one of the major spin-offs of electronic technology. Thaddeus Cahill's "Telharmonium" made use of early electro-magnetic techniques to transmit music down the telephone lines, but even before that time, demonstrating the "singing (electric) arc" had been a popular parlour game. Instruments such as the "Trautonium" and the "Ondes Martenot" made use of valve technology to produce sounds controlled by a piano-type keyboard, but it was Laurens Hammond's Hammond Organ which really established the popular concept of an electronic musical instrument.

However, many musicians and designers were frustrated by the limitations of the Hammond Organ, and one — Dr. Robert Moog — found the answer in 1964. In that year, the young engineer read his paper on "Voltage Controlled Electronic Music Modules" to the Audio Engineering Society in the USA — and the rest is history.

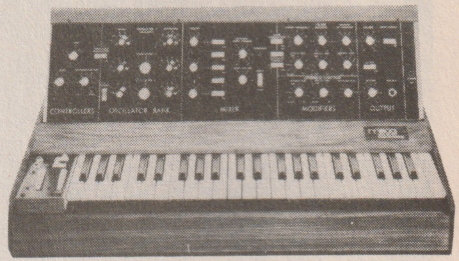
Early modular synthesisers tended to be on the complex side...



A Là Moog

Moog's idea was to build an oscillator (in this application, a basic sound-producing circuit) which could have its pitch very accurately varied by an input voltage. Quite arbitrarily, he decided to derive this voltage from a piano-type keyboard (simply because this is more similar to a row of electronic switches than, say, a guitar's fretboard or a flute), and so 1965 saw the birth of the Moog Synthesizer, a name which became generic for all makes of synths.

Apart from the Voltage Controlled Oscillator, Moog designed a Voltage Controlled Filter and a Voltage Controlled Amplifier. The VCF was designed to modify the sound from an oscillator by cutting out high overtones (low-pass mode), low tones (high-pass mode) or certain harmonics only (band-pass mode). In this way synthesizers could imitate the tones of flutes, oboes, bassoons and so on.



The MiniMoog was the first successful pre-patch synth.

The VCA, when controlled by an Envelope Generator circuit, "shapes" the volume of the sounds to more closely resemble certain instruments. For instance, the sound of a piano begins suddenly but fades out slowly, whereas the sound of a violin takes a little while to build up to full volume but can then stop almost instantaneously. Moog's early synthesizers could produce some reasonable imitations, although the best of the early exponents (such as Walter Carlos) preferred to use them to create many new sounds as well.

Moog's early synths were modular — that is, each element of the synth came in a separate unit which had to be connected to others with patch leads.

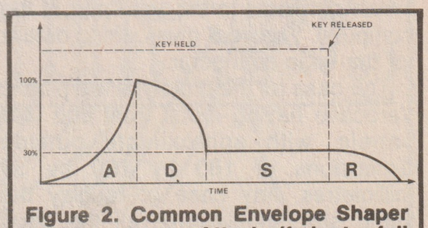


Figure 2. Common Envelope Shaper parameters are Attack (fade to full volume), Decay (fade down to sustain), Sustain (hold volume) and Release (fade after the key is released). Hence ADSR. In complex synths and most FM designs this is ADBSSR (Attack, Decay, Break Point, Slope, Sustain, Release).

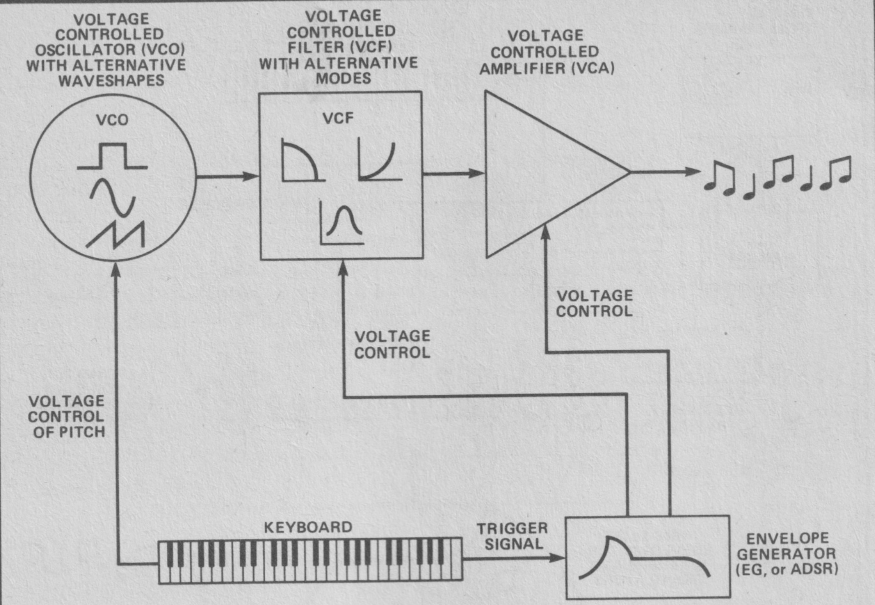

VOLTAGE CONTROLLED OSCILLATOR (VCO) WITH ALTERNATIVE WAVESHAPES
VOLTAGE CONTROLLED FILTER (VCF) WITH ALTERNATIVE MODES
VOLTAGE CONTROLLED AMPLIFIER (VCA)
VCO
VCF
VOLTAGE CONTROL
VOLTAGE CONTROL
VOLTAGE CONTROL OF PITCH
KEYBOARD
TRIGGER SIGNAL
ENVELOPE GENERATOR (EG, or ADSR)

Figure 1: Waveforms from the VCO are treated by the VCF to vary the sound. The signal then goes through the VCA. An Envelope Shaper varies the setting of the VCF and VCA during each note to simulate the sound of an organ, for instance, or to create a new sound.

This wasn't the fastest way of working, and the Moog synth only became really popular when it was packaged in the compact format of the MiniMoog, which was popularized by Rick Wakeman, Keith Emerson and others. Although slightly limited compared to the large modular systems, the ready-patched MiniMoog remains relatively fast to use and very expressive in performance.

But synths such as the MiniMoog, the ARP "Odyssey" and the EMS "Synthi A" still had many drawbacks in terms of setting up new sounds. It could take many hours to perfect the sound you wanted, and it was often impossible to reproduce the effect at a later date. Also, most synths remained stubbornly monophonic — unlike the Hammond organ, they could only play one note at a time, having only one or at best two or three oscillators.

Early attempts to solve these problems, such as on the Yamaha CS80 and the PolyMoog, were only partially successful. It took the advent of micro technology to produce the next step in synth design, and ironically enough this was inspired by designer Dave Smith's desire to build a polyphonic MiniMoog.

Smith found that a microprocessor could allow his synth keyboard to juggle with just five pairs of oscillators, giving him the ability to play chords without having to build an oscillator circuit for every key. His design, the Sequential Circuit Prophet 5, also made the processor look at every setting of the filter, oscillator and amplifier and store those settings away in Random Access Memory in complete sets, or "patches" (so called despite the fact that there wasn't a patched lead in sight!).

Analog Vs. Digital

So, at a stroke, the synthesizer became both polyphonic and programmable in terms of the sounds available. The development of synth technology then took a bit of a break while other manufacturers (such as Roland, Yamaha and ARP) brought out their own polyphonic programmable synths. All of these shared the time-honoured Moog system of voltage-controlled oscillators, filters and amplifiers, collectively referred to as analog design.

But development was inevitable, and the Japanese company Yamaha eventually ditched their analog synth range in favour of a new, digital design based on FM (Frequency Modulation) technology.

FM differs from analog sound production, in a number of ways. Analog synths start with a basic oscillator, and filter its tone down to produce new sounds. This is called subtractive synthesis, and obviously has disadvantages because you can only take away from a basic sound rather than build on it. Contrastingly, the FM system starts with very simple sounds (sine waves in fact) and combines them to form any desired complex sound.

The FM system has its own terminology which still baffles users today, three or four years after its introduction, and the Yamaha synths

Roland
JUPITER-6

Polyphonic analogue synths like the Jupiter 6 are now relatively easy to use.

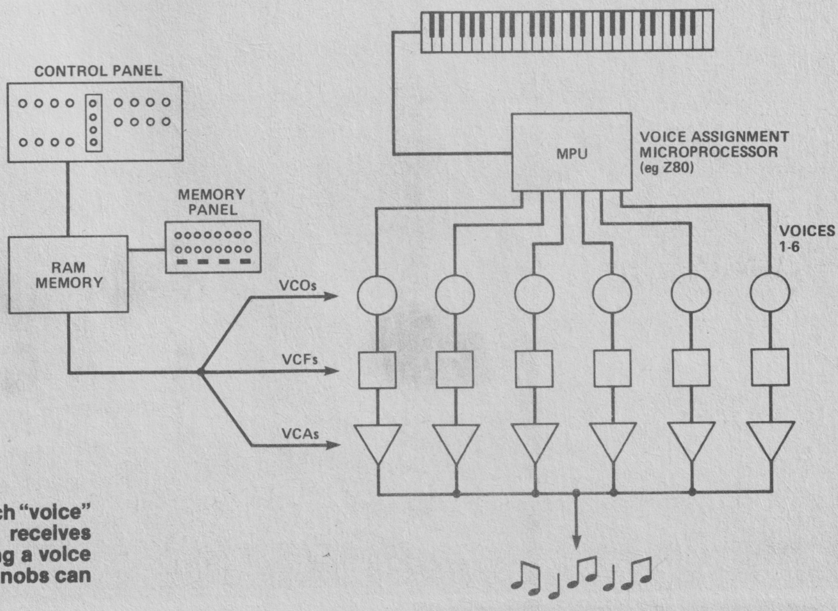
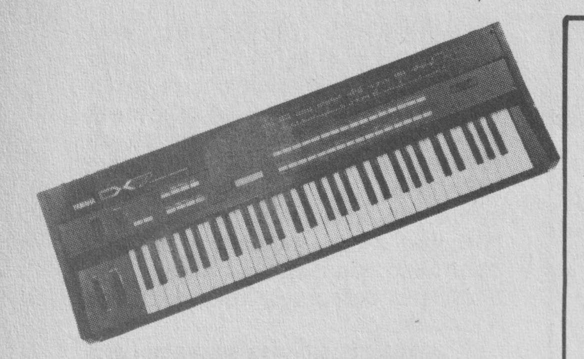

CONTROL PANEL
MEMORY PANEL
RAM MEMORY
MPU
VOICE ASSIGNMENT MICROPROCESSOR (eg Z80)
VOICES 1-6
VCOs
VCFs
VCAs

Figure 3. A typical polyphone analogue synth: each "voice" is controlled from a microprocessor which receives instructions from the keyboard, instead of having a voice under every key. The settings from the control knobs can be stored and recalled.



Yamaha's DX7 made FM synthesis a success.

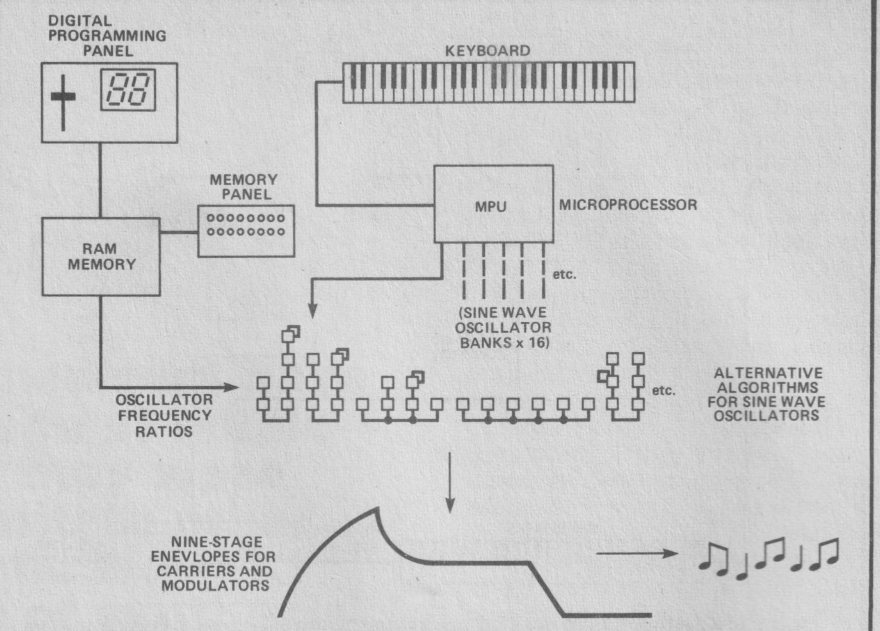
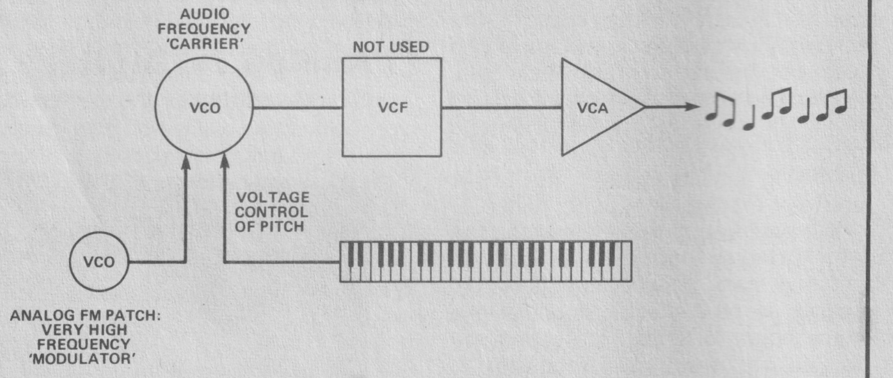

DIGITAL PROGRAMMING PANEL
KEYBOARD
MEMORY PANEL
MPU
MICROPROCESSOR
RAM MEMORY
etc.
(SINE WAVE OSCILLATOR BANKS x 16)
OSCILLATOR FREQUENCY RATIOS
etc.
ALTERNATIVE ALGORITHMS FOR SINE WAVE OSCILLATORS
NINE-STAGE ENEVLOPES FOR CARRIERS AND MODULATORS

Figure 4 (a) A typical FM synthesiser. The sine wave oscillator banks are used in different arrangements or 'Algorithms' to produce different types of effect; the amplifier circuit and keyboard are quite familiar from older designs. (b) An experimental analog FM patch: you can create a single FM voice on a modular analog synth using the patch below.


AUDIO FREQUENCY 'CARRIER'
NOT USED
VCO
VCF
VCA
VOLTAGE CONTROL OF PITCH
VCO
ANALOG FM PATCH: VERY HIGH FREQUENCY 'MODULATOR'

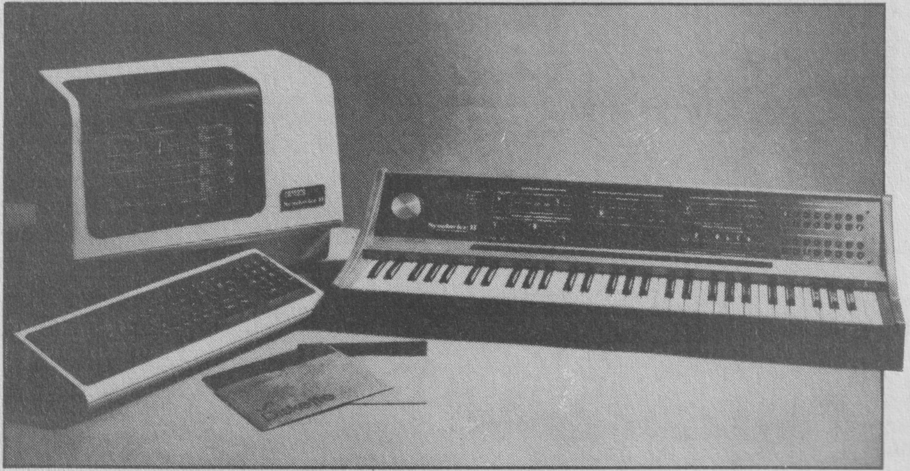
need powerful processors to deal with the hundreds of parameters (typically 147 on the Yamaha DX7) which go to define each sound. FM sine wave oscillators can act in either of two ways; when acting as a "Carrier" they actually produce a sound which is audible, and when acting as a "Modulator" they affect the pitch of a Carrier at a variable speed. The exact assignment of these tasks to each of the sine wave bands is called an "Algorithm", and the exact pitches and frequency ratios chosen for the Carriers and Modulators determine the overtones produced, and so the final sound.

So, on FM synths there's not a filter in sight, although Yamaha's DX range synths do have an amplifier circuit, but one considerably more complicated than those on most analog designs. FM sounds are typically bright, buzzy and clinical compared to warmer analog effects, and while there are plenty of designs about for home-built analog synths (either for individual oscillator, filter and other modules or for complete instruments), you won't find much in the way of do-it-yourself FM. The reason is that Yamaha custom-build chips for their FM synths, and this can only be done under license from the patent-holders of FM Synthesis, Dr. John Chowning and the New England Digital Corporation, manufacturers of the huge Synclavier FM computer synth. You can, however, make a few simple experiments with FM, as we'll see.

But on the subject of computer synths, there's a new type of synthesizer on the market which is just becoming economically accessible — it's the polyphonic Sound Sampler.

The basic problem of synthesis had always been to create full, active sounds similar to those of "real" acoustic instruments. We may not always want the exact sound of a piano, a trumpet or a violin, but it would be nice to have something with as much life and energy, rather than a simple square wave buzz.

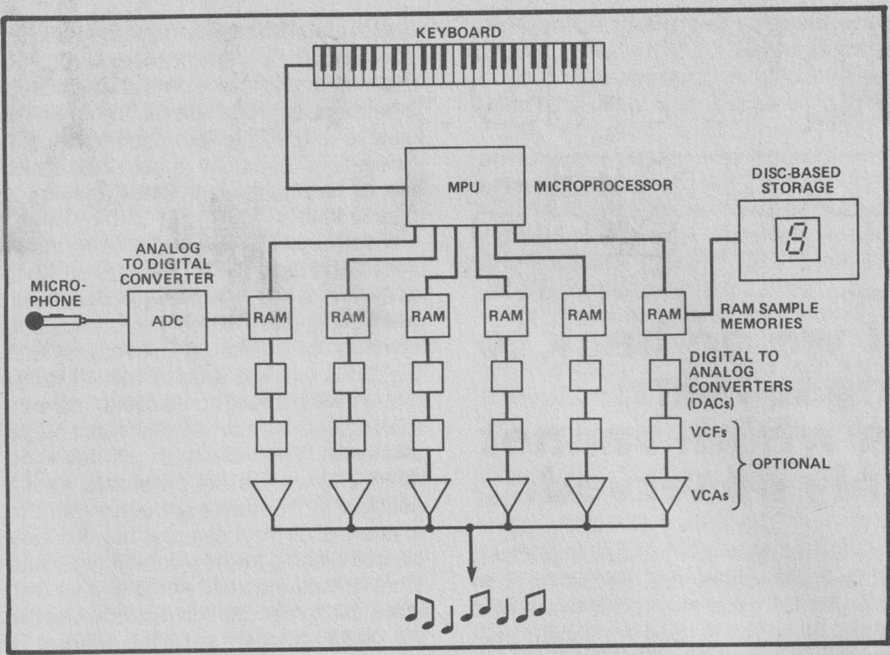
The computerised Synclavier combines FM synthesis, and sampling of real sounds.

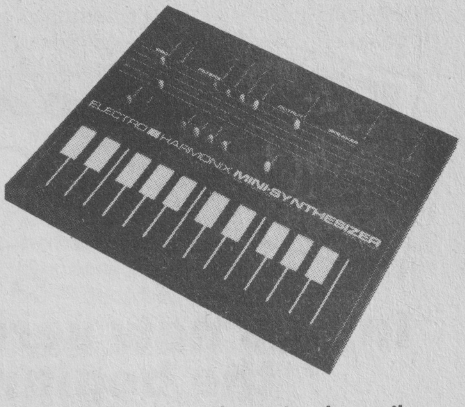


Sampling

FM is one possible solution, but another answer is to digitally record an existing instrument's sound in RAM memory and play it back from a keyboard. Playing the sound back at different clock rates alters its pitch so you can play different notes, and providing several slices of memory (allocated to the keyboard by a microprocessor as on conventional synths) allows you to play chords using any sound you'd care to "sample".

Just imagine it — you can hire a trumpet player, record one note of his playing, and then reproduce vast chords of trumpets from the keyboard. You can


KEYBOARD
MPU
MICROPROCESSOR
DISC-BASED STORAGE
ANALOG TO DIGITAL CONVERTER
MICRO-PHONE
ADC
RAM
RAM
RAM
RAM
RAM
RAM
RAM SAMPLE MEMORIES
DIGITAL TO ANALOG CONVERTERS (DACs)
VCFs
OPTIONAL
VCAs


ELECTRO HARMONIX MINI-SYNTHESIZER

(Above) A basic synth can teach you the principles of synthesis. Figure 5 (left) A sampling synth. This has a set of digital memories for real sounds, which may or may not be matched with more familiar synthesiser circuits.

TONY BENYON

"It's the usual line up — bass, drums, lead and computer programmer."
An antique from 1971.

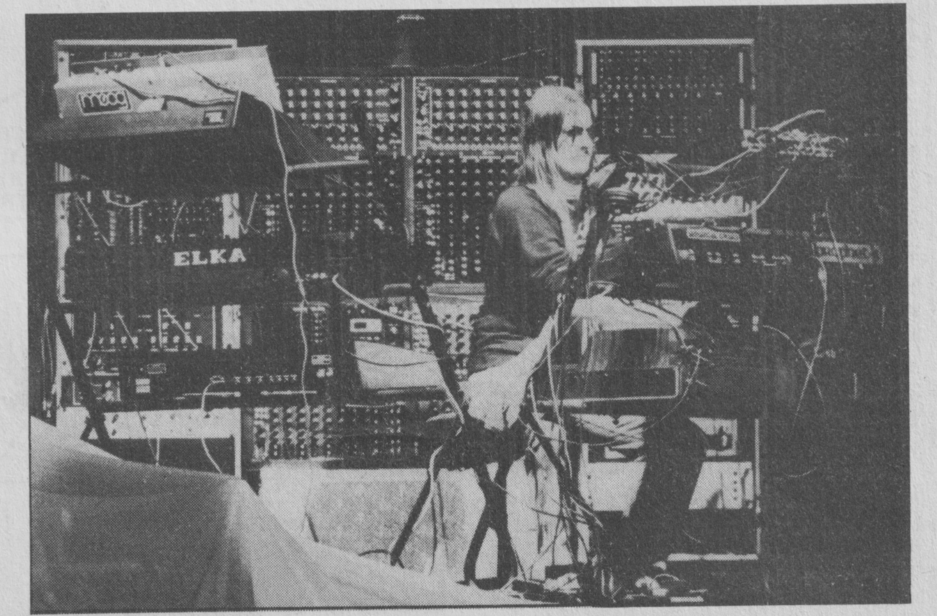
Thanks to the original lone groover Tony Benyon for permission to reproduce this cartoon.

do the same with voices, guitars, in fact any instrument. More excitingly, you can sample non-musical sounds such as breaking glass or animal sounds, and synthesizer effects which are too complicated to reproduce, and work them into any piece of music instead of conventional drum or string or brass sounds.

Sound sampling has already evolved a sonic vocabulary of its own after having been used in professional studios having the Australian Fairlight Computer Musical Instrument, the Synclavier with sampling option, or the Emulator. Paul Hardcastle's "19", which recently reached the top of the charts, was largely based on sound samples taken from a TV programme using an Emulator II.

The Emulator II, the new Ensoniq Mirage and the PPG Waveterm all combine sampling capabilities with many of the features of familiar analog synths. That is, they have filters, amplifiers and envelope shapers which can alter the sampled sound and use it as basis for building up very complex sounds which are totally unfamiliar.

So synthesizer design has come full circle, from imitating acoustic sounds with analog circuits to digitally recording the real thing and carefully working on the sounds with those same circuits. Physically, synthesizers have changed out of all recognition, but internally much remains the same. Learning first principles on a very basic synth, or better still buildiing some of your own circuits, will stand you in good stead if you want to go on to learn more about the most advanced synthesizers available today.



Compact Disc star Christophe Franke of Tangerine Dream tries to extract his chewing gum from a Prophet 5, the first microprocessor controlled, programmable polyphonic synthesiser.

FROM ATOMS TO AMPERES

F.A. Wilson

In this new serial we go back to the beginning to learn about phenomena which cannot be explained, only observed.

THOSE WHO get wise to the fundamentals of electricity become the best placed for understanding what it can do. No worries if you have forgotton your school physics: we will revise what is needed first. Mathematics? Of course, but nothing more complicated than simple equations and a bit of trigonometry. A calculator is handy, but not essential.

About The Series

We are going to look at the very essence of life, electricity. There are high-level text books for physicists and sketchy abstracts for engineers; we are going to attempt to fill the gap in between.

Equipped with a better than average intimacy with the fundamentals of electricity the reader should gain a lifetime of help with future studies. The series presumes little prior scientific knowledge and only requires a modicum of agility with arithmetic. The level of mathematics has been kept deliberately low, so a moderate knowledge of school algebra and a little basic trigonometry should see us through. We will brush up on what should have been remembered from school first. It can be confidently said that to most of us the subject can be fascinating and absorbing, and even impart a more enlightened outlook on the world we live in.

As with most texts, the series is sprinkled with calculations to add realism to the ideas. There may be some who are in love with their calculators but there are many who are not: the latter can be assured that skipping over the arithmetic and noting the answers will only detract a very little from the main message.

Now we must first give credit to those scientists to whom we are indebted for the benefits of electricity and electronics. They were not men who got up in the morning and just let it all happen: they wanted to know why and how it happened (whatever it was). From their work and published papers others have been able to design the hardware which surrounds us today, from street lighting and electric trains to circuits for experimenters.Newton and Faraday spring to mind immediately but there are many more. We will meet some as we progress.

Avoiding The Deep End

In the early days, when lightning lit up the skies, mortals down below quaked in terror. Electricity was then the enemy of man. Yet as the years passed it was discovered in other forms, tamed and its enormous power harnessed. But to do this man had to learn some of electricity's innermost secrets. Not all, for nature is elusive and still keeps much to herself. Nowadays electricity is both friend and foe, a helpful servant which can be a deadly killer.

Mostly we see electricity in terms of volts and amperes. Behind all this, however, is the content of matter itself, everything, even ourselves, comprising myriads of miniature universes with electrons whirling around within atoms, just as the planets hurtle round the sun. Electricity is the prime constituent of our existence. What a pity it can never be seen! But we are not physicists whose mission in life is to dig deeply into the mysteries of the universe, but ordinary people wishing to know something about what makes electricity tick. It is as well therefore to decide first just how deeply we wish to get involved. There is certainly no lack of information available but much of it requires an above-average expertise in maths so we must keep things relatively simple. Also we need to look more carefully at some of the concepts we may have accepted without much thought in the past, for the more we find out about electricity, the more shaky some of our previous ideas

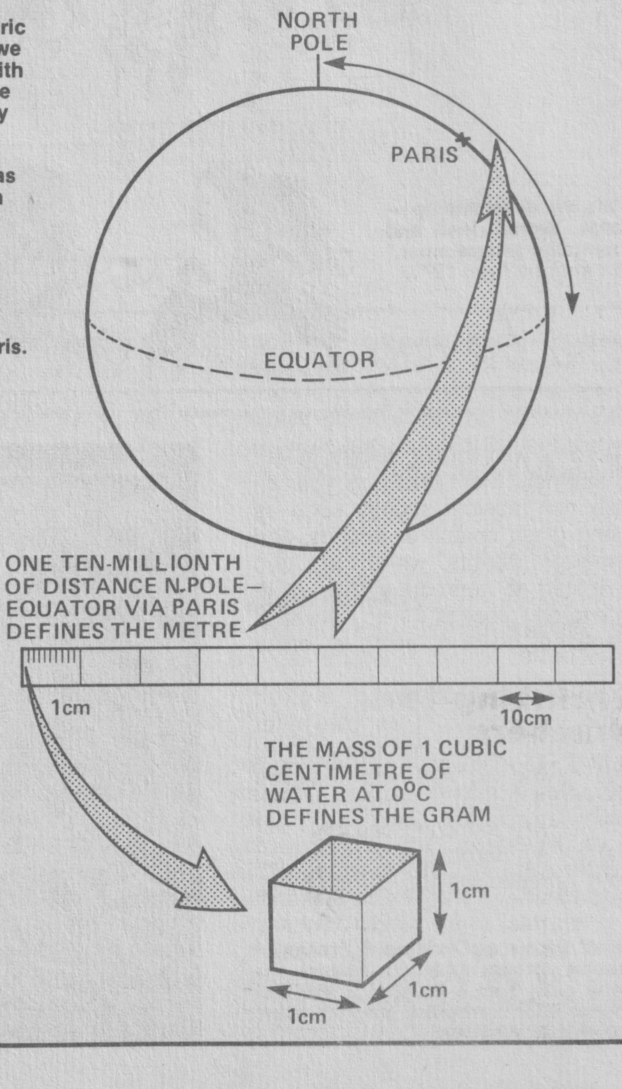


Figure 1. The metric system — which we will be working with in months to come — was adopted by the French Academy of Science in 1971, as part of the French Revolution. The metre was calculated to be 1/10Mth of the distance from the North Pole to the Equator — via Paris.

NORTH POLE
PARIS
EQUATOR
ONE TEN-MILLIONTH OF DISTANCE N.POLE–EQUATOR VIA PARIS DEFINES THE METRE
1cm
10cm
THE MASS OF 1 CUBIC CENTIMETRE OF WATER AT 0°C DEFINES THE GRAM
1cm
1cm
1cm

become. It is therefore important that we do not gloss over things on the assumption that we understand them when in fact we do not. An illustration of this is given by a look at the general idea of 'space'.

First thoughts are that space is what we see on looking up into a clear sky: the vacuum beyond the Earth's atmosphere. Yet through space astronauts communicate with the ground by radio, stars emit radiation we receive on earth, including the sun. Then there are the gravitational and magnetic fields of Jupiter, Neptune, Saturn, and the rest which astrologers tell us shape our characters. Unquestionably all these influences and many more are present in space, so perhaps it is not so empty after all.

What this indicates is that when we think deeply about aspects of Nature, uncertainties creep in. We must leave the more complicated puzzles to the physicists, relying on the fact that our knowledge of the atomic world can be greatly improved even if we only study it at a modest level.

Now this is hardly an encouraging starts, exposing our ignorance of Nature's box of tricks! However, by observation and experiment over the centuries scientists have managed to set down many of the **rules or laws** by which Nature operates. She may keep us in the dark as to her motives but she cannot stop us from prying. But when we do, we cannot see for ourselves what goes on, so that the pictures we draw must be accepted as intelligent but inadequate guesswork. Nevertheless such pictures are becoming clearer, and continual experimentation tends to confirm that we are on the right track. This can be summed up by saying that phenomena such as **force, charge, field** and **gravitation** must be accepted as what Nature has provided, and even though we cannot clearly describe what they are, we do know a great deal about what they do. Without this practical knowledge modern developments could never come to fruition. Keeping this always in mind, we can become more friendly with the atom, and its hard-working assistant, the electron, but first we must study the laws pertaining to the various items in Nature's tool kit. These laws will tell us how and why the electron acts.

Shrinking The Numbers

Not many millionaires read *Electronics Monthly*, because there are not many of them about. To most of us even one thousand pounds is a lot of money. Imagine one thousand packets, each containing one thousand pounds in say, ten rows of one hundred packets each. That is one million. A pleasant thought. Human beings can handle such numbers — just. Larger numbers tend to get difficult. Who can truly grasp the significance of the World population at around 5,000,000,000? As the number of noughts grows so does our confusion. Scientists long ago decided that all these noughts everywhere made arithmetic unwieldy and so was born **scientific notation**. With this, we can work easily with extremely large or small numbers, which is especially useful when we are trying to see what the various formulae have to tell us.

Let us see the problem as it affects us here by taking a number which in physics crops up time and time again (believe it or not). Written 'longhand' it is:

0.000,000,000,000,000,000,160,2

(the electron charge — more about this later) Some other numbers are twice this length. Scientific notation is not only a method of writing the number down in a shortened form but it also allows us to calculate with it directly. Hence, the difficulty of writing down a string of noughts correctly no longer exists. A little revision, first, however:

$10 \times 10 = 100$ or 10^2 (ten squared, or to the power of 2).
$10 \times 10 \times 10 = 1000$ or 10^3 (ten cubed or to the power of 3).
$1000 \times 1000 = 10^3 \times 10^3 = 1{,}000{,}000$ or 10^6 (ten to the sixth, or to the power of 6).

so we can express multiples of 10 as 'powers of ten' using the little raised number or **exponent** (from Latin: 'placed outside') to indicate the power. Thus 10^9 stands for 1,000,000,000, which is 10 multiplied by itself 9 times. Note also that $10^1 = 10$ and $10^0 = 1$.

Multiplying by 10 increases the exponent by 1, multiplying by 100 (10^2) increases it by 2, etc. from which comes the general rule:

$$10^x \times 10^y = 10^{(x+y)}$$

Similarly, dividing by 10 decreases the exponent by 1, hence:

$$10^x \div 10^y = 10^{(x-y)}$$

Let us test this out by multiplying 10,000 by 100.

$$10{,}000 = 10^4 \qquad 100 = 10^2$$
$$\therefore 10^4 \times 10^2 = 10^{4+2} = 10^6 \text{ (one million).}$$

Conversely, dividing 1,000,000 by 100 gives:

$$10^6 \div 10^2 = 10^{6-2} = 10^4$$
$$\text{(or equally } 10^6 \times 10^{-2} = 10^{6+(-2)} = 10^4\text{)}$$

so we move from division to multiplication by changing the sign of the exponent.

Now, with numbers containing a decimal point, multiplying by 10 merely shifts the point one place to the right, and dividing by 10 shifts it one place to the left, so 16.02 multiplied by 100 becomes 1602, (ie $16.02 \times 10^2 = 1602$); 16.02 divided by $10^2 = 0.1602$. In the same way $16.02 \times 10^{-4} = 0.001602$. We are simply shifting decimal points and changing the exponent accordingly.

This is what scientific notation does: the world population of 5,000,000,000 becomes simply 5×10^9 and the almost unmanageable number for the electron becomes 16.02×10^{-20}. *(Easy — Ed.)*

Calculator Language

However, so that we all adopt the same technique, the number is usually **normalized**, meaning that only one digit precedes the decimal point, so 16.02×10^{-20} becomes 1.602×10^{-19}.

A few other examples of numbers quoted in scientific notation are:

$$854 = 8.54 \times 10^2$$
$$2847.3 = 2.8473 \times 10^3$$
$$0.00039 = 3.9 \times 10^{-4}$$

Now this is where scientific calculators and home computers come in. Most handle scientific notation directly. However, they cannot display exponents in the form of raised numbers but do so in other ways. For example 1.602×10^{-19} might be displayed as 1.602 -19 or as 1.602 E-19. The user manual must be consulted for the actual display arrangements and method of use. Generally, entries not in normalized form will be automatically changed. To take a single example:

$$\frac{1386.72 \times 10^{11}}{0.0642 \times 10^{-7}}$$

is calculated directly by scientific calculator or computer to give the result 2.16×10^{22}. However if you are only in possession of an ordinary pocket calculator, don't despair, all that is needed is a little mental arithmetic to take care of the exponents for:

$$\frac{1386.72 \times 10^{11}}{0.0642 \times 10^{-7}} = \frac{1386.72}{0.0642} \times 10^{18}$$

$$\left(\frac{10^{11}}{10^{-7}} = 10^{11} \times 10^{7} = 10^{18}\right)$$

Use the calculator for the nasty looking fraction to get 21600.

The final result is therefore 21600×10^{18}, which becomes 2.16×10^{22} on normalizing. We have simply handled the powers of ten separately, remembering the rule that shifting from numerator to denominator (or vice versa) changes the sign of the exponent.

As we become more and more interested in things atomic such fantastically small and large numbers will appear more frequently. It is pleasantly surprising to find that although one cannot have any conception of their physical representation, they begin to make sense with practice and in fact we then come to think directly in scientific notation. There will be plenty of exercise in this technique later.

Next month we shall be describing the standard units and abbreviations used in Physics, and revising some useful fundamental principles.

VIBRATION AMPLIFIER FOR MOTOR TESTING

DUNLOP
'Long' Cycle Tyre
Repair Outfit

What's the rattle? Put this to your ear, and you may find out.

I. Rees

THE OLD, OLD method of using a long screwdriver as a makeshift amplifier and earpiece for listening to engine noises was convenient, but somewhat limited, quite apart from any comments which might arise if one was found listening to a screwdriver.

The Vibration Amplifier described in this article provides a convenient and slightly more sophisticated means of listening to sounds within a running mechanism. It involves the use of a pair of headphones to convert the amplifier into a stethoscope-like instrument. With the help of this device, out-of-place noises in an engine can be pinned down before extensive dismantling, possibly in the wrong part of the machine, has been undertaken.

The Circuit

Figure 1 shows the circuit diagram. This comprises, mainly, of a transducer and an amplifier feeding a pair of headphones.

The transducer is a microphone of the electret type widely found in cassette recorders. Normally, airborne noise would swamp conducted sounds, but a piece of PVC tape placed across the microphone's small inlet hole removes this. Airborne sounds are almost totally eliminated.

The electret is not affected by magnetic fields, and can be used in mains driven units where 50Hz induction would otherwise be a problem.

Vibrations are transmitted up the probe (**Figure 2**) to the plate which has the microphone attached to it. The mic picks up the sound through its body.

Bias for the microphone is supplied through R1. The output is coupled through C2 to the preset gain control, RV1. IC1 amplifies the signal two hundred times before supplying its output to the stereo jack socket SK1.

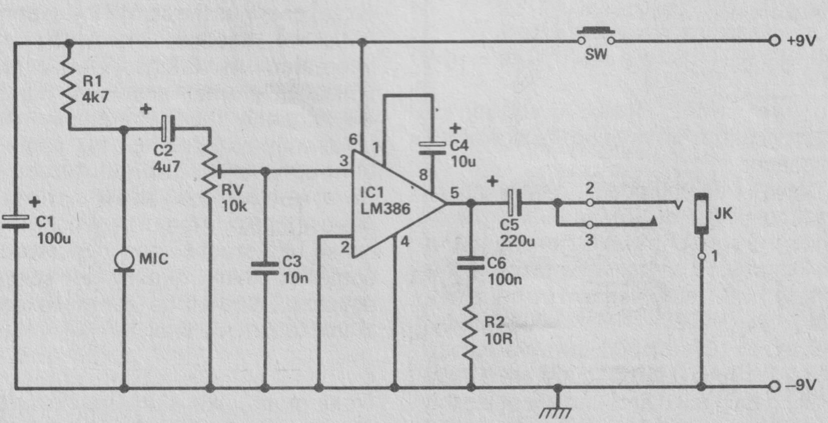


Figure 1. The circuit comprises mainly a transducer to convert the vibration to a signal, and an amplifier.

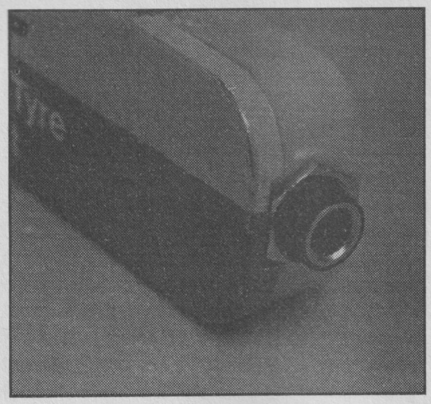
The jack socket is a standard ¼in type. Although rather large, it allows the use of ordinary hifi headphones, which gives an extended bass response. If they are the type with 'earmuff' pads they will give further rejection of airborne noise.

The amplifier IC1 is an LM386. This was chosen for its small size, high gain and requires few external supporting components.

A push button switch (SW1) allows the PP3 battery supply only to be switched on while the unit is in use.

Construction

The amplifier, including its battery, is contained in a Dunlop cycle puncture repair kit box, which apparently led the EM editorial team to decide that this was an automatic puncture device, before reading the article. The bottom of the box is plastic with a hinged metal lid. A 16SWG aluminium chassis (**Figure 3**) strengthens the bottom of the box for



Parts list

RESISTORS

R1	4k7 1/8W 5%
R2	10R 1/8W 5%
RV1	10k sub-min horiz preset

CAPACITORS

C1	100u10V electro
C2	4u7 63V electro
C3	10n 50V electro
C4	10u 35V electro
C5	220u 16V electro
C6	100n 50V ceramic

MISCELLANEOUS

SW1	single pole push switch normally open
SK1	¼in stereo jack socket

Electret microphone insert; PP3 battery connector; 0.1in matrix board, 32 × 32mm; case to suit.

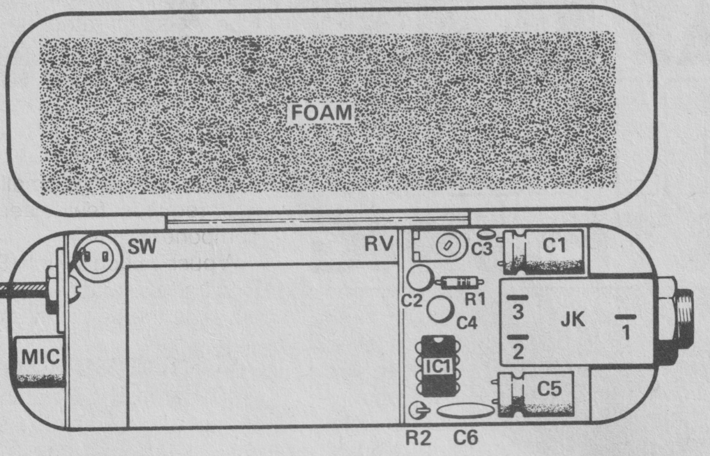


Figure 2. The general interal layout of the Vibration Amplifier. The foam holds the works firmly in place.

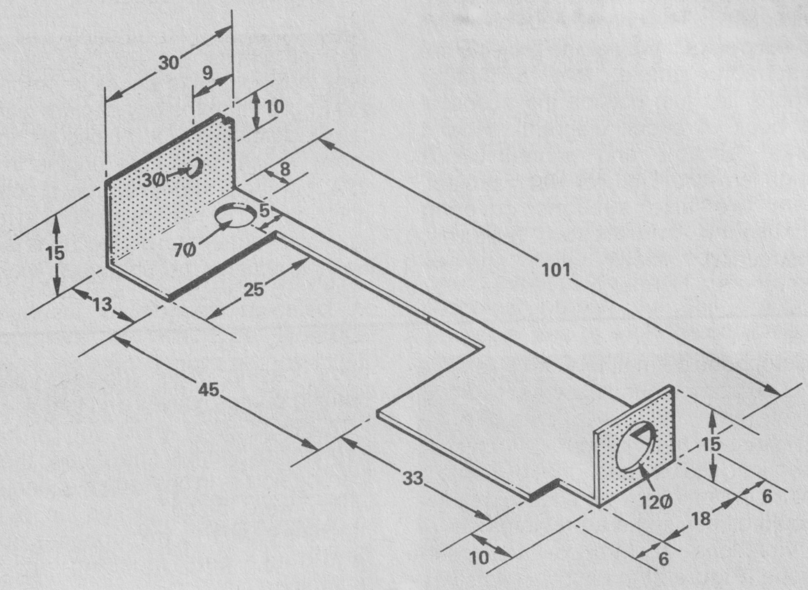


Figure 3. Measurements for an internal aluminium chassis for the case as used.

mounting the probe and jack socket. The on/off switch protrudes through the lower part of the case, with its fixing nut on the outside.

The microphone is positioned between the case and the chassis. The terminals of the mic are against the wall of the case, with the PVC covered vent facing the chassis upright to pick up the vibrations.

The probe is a 6mm screw, 30mm long, sharpened and fixed to the same upright. A solder tag under the head of the probe connects the lid and chassis to the -ve line. The circuit is wired to a small piece of matrix board 32 × 32mm (**Figure 4**), shaped to fit tightly around the jack socket. A piece of stout card fits under the board and flaps up on the battery side to prevent short circuits.

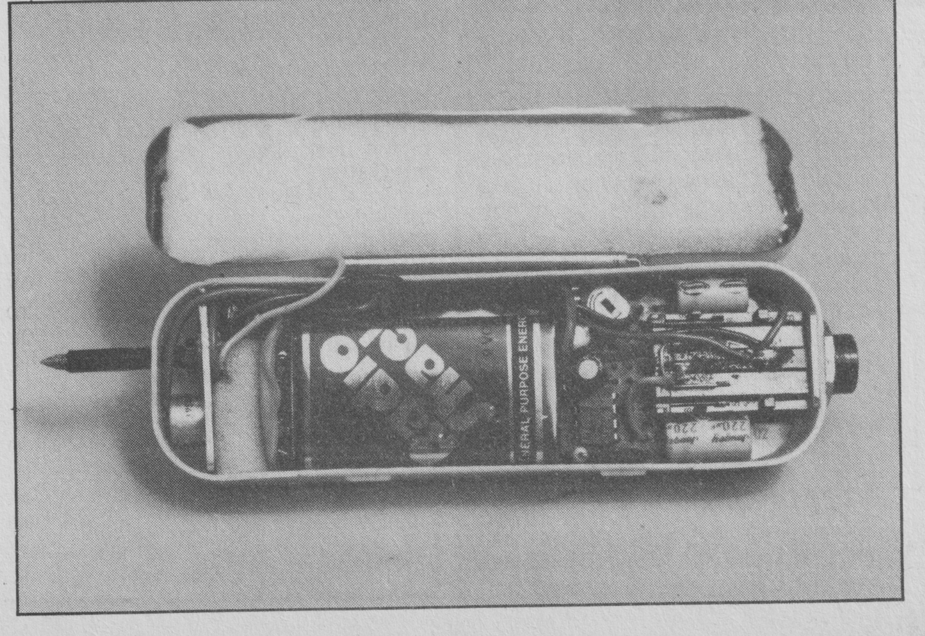
The jack nut is fitted to the outside of the case. A small niche needs to be cut out of the lid to allow the top to close over the jack nut. The chassis is firmly held by the jack and switch fixings.

The matrix board is not retained other than by its jigsaw fit around the jack socket. However, a thin strip of sponge is glued in the lid to retain any upward movement of the battery or board.

I broke a plastic lug off the lid hinge and, as a result, had to replace it with a paperclip version, similar to old fashioned wire hinges as supplied in pre-plastic days. The result was a much improved hinge: I include this piece of information for the perfectionists.

Operation

With a suitable pair of 8ohm headphones connected, and the gain control RV1 at maximum, touch the end of the probe with your finger. You should hear a sound similar to that found when you touch the stylus of a record player with the volume up Little or no spoken sound should be heard through the



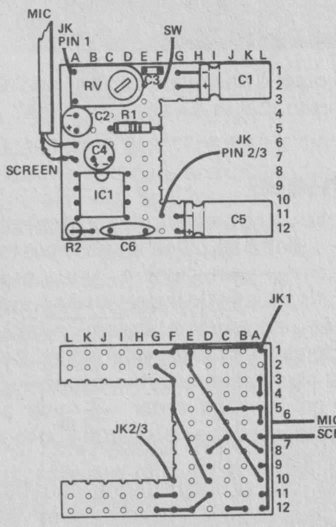


Figure 4. The layout for the stripboard, with the back-of-board wiring.

COVER VERSION

Electronic Projects For Musicians. Craig Anderton, Amsco Publications, £9.95. Reviewer: R. A. Penfold.

Although rather expensive by electronic project book standards, with some 220 pages of slightly more than A4 size, this American book could not really be accurately described as over-priced in simple page area per pound terms. A unique feature (as far as I am aware) is the monophonic demonstration disc supplied free with each copy. This is only of the usual very thin and flexible give-away type, but it is sufficient to demonstrate well the capabilities of the effects units described in the book, and is certainly a worthwhile inclusion. I approached playing this square disc with some trepidation, but there were no problems!

The book is primarily aimed at the musician who wishes to build some low cost equipment, rather than at the electronics hobbyist, and a substantial section at the beginning of the book is taken up with topics such as component identification and resistor colour codes. There are further sections covering mechanical aspects of construction, and soldering techniques. This should certainly be of great help to beginners, but obviously means that anyone with some experience of project construction is paying for pages they do not need. There is a useful section at the back of the book which deals with using and getting the most from completed projects, but the bulk of the book is taken up with twenty seven sections which deal with a variety of music projects.

Many of the projects are effects units, but other topics such as mixers, a noise gate, guitar rewiring, a dual power supply, a headphone amplifier, making patch cords, and bypassing are included. Most popular types of effects unit are described, including several types of fuzz, a phaser, and other types of filter effect. The only obvious omission is a flanger, but most of the projects are reasonably simple, and a flanger perhaps lies just outside the scope of this book. A circuit diagram, "How it Works" section, and printed circuit design are provided for each project. There are also sections covering background information, features, construction, "difficulty rating", and any necessary setting up procedures.

Usefully, sections on modifications and fault finding hints are also provided, together with specifications where appropriate. Anyone who normally only uses ready-made printed circuit boards would have to learn the art of using a PCB pen and ferric chloride, but the board designs do not look to be very complex in the main. The circuits could probably be built on stripboard with little difficulty.

The projects all appear to be practical types which should function well, and the instructions are well laid out and easy to follow. Differences in American and English terminology can sometimes cause difficulties when using technical books which have their origins in the USA, but there should be few problems in this case. The really big question mark against my American project book is can the components be obtained? Most of

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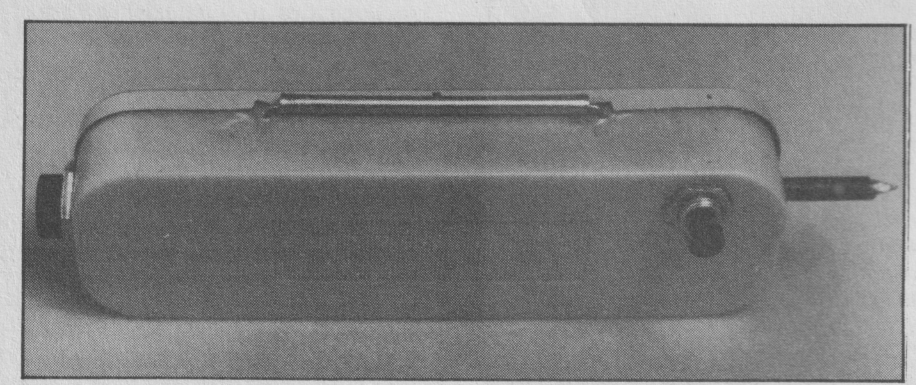

headphones. The device can be used as a diagnistic tool in a variety of trades.

Vehicle

If the tip of the probe is placed against a metal surface as close as possible to the moving part which you are investigating. Listen to the relative sound levels of extraneous noise versus engine noise; in this way you can locate the most likely source of the sound.

Bearings in good order will give a smooth hiss when the tip is placed on the outer surface.

Tappits out of adjustment can be located before the rocker cover is removed, by moving the probe along the top.

You can listen to the wheel bearings by jacking up the vehicle and spinning the wheels. Main bearings, the generator, the water and petrol pumps can all be listened to in a similar way.

The sounds revealed by the vibration amplifier can be judged by experience. If the source of the noise sounds rough, a judgement can be made as to the condition of the bearing, etc.

A small amount of ignition noise will be picked up by the circuit when working near the spark plugs. This can be useful for detecting misfire by holding the unit hear to (but not touching) each plug in turn and checking the clicks for regularity.

Electrical

Electric motors can be inspected for bearing or brush wear by applying the probe tips to appropriate parts of the body casing. This includes central heating pumps, valves and thermostats. If it slides, clicks or whirrs, you can hear it through the phones. I have used the unit to trace noises in cassette mechanisms, which were being picked up by the machine's internal microphones during recording, and messing up the recordings.

Resonances in loudspeaker cabinets can be traced by touching the probe and listening to the amplitude nodes. If the vibration level is too high, reduce the gain by adjusting RV1 to a convenient volume.

The suggestions above are far from exhaustive, and the applications are only limited by the needs and ingenuity of the user. (Death Watch Beetle, for instance?)

Shop

All the components, including an electret mic insert (solder your own lead on) are available from Maplin (0702 554155) apart from the IC, which is available from Tandy (Cat. No. 276-1731) or Watford (0923 37774). The unit should cost about £5 excluding case and the 6mm screw. If you are using junkbox or recycled components, the cost should be considerably cheaper.

Hom
Recor
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Electronic
Projects
for Musicians.

the projects are based on ordinary operational amplifiers and dual or quad operational amplifier devices. The low noise types specified might prove to be elusive, but the suggested alternatives are standard, easy to obtain types, which should give a high level of performance. The main problem is likely to be the opto-isolator which is used in some of the more interesting projects (the phaser for example), which is a type having a LED driving a photoresistor.

Searching through several component catalogues revealed no component of this type whatever (more's the pity). It might be possible to improvise something from an ordinary LED plus a cadmium sulphide cell, or it might be possible to import the correct part from one of the suppliers listed in the book. Neither approach is very satisfactory though, and this has to be regarded as something of a black mark against this otherwise excellent publication. Most of the other parts seem to be readily available apart from a couple of regulator chips and one or two transformers which might cause a few headaches.

With some of the projects effectively deleted it still represents an interesting and useful book, but obviously represents relatively poor value for money. An experienced constructor could probably manage to overcome these difficulties, but would be buying a lot of unnecessary background information, and would again be getting compromised value for money. If you would like to build some music projects and you can afford this book it still probably represents a worthwhile investment.

Studio Recording For Musicians. Fred Miller, Amsco Publications, £9.95. Revewer: Ron Keely.

Despite the fact that it is an American publication, this book is an excellent guide to recording studio theory and practice. It will be a good introduction to the business for anyone with ambitions to break into the competitive field of studio engineering, or for a musician who doesn't like being baffled by the technological wizardry of the modern recording studio.

Studio Recording For Musicians begins with an outline of the participants in a recording session — what they do and what they are called. Next the basic studio equipment is described, from microphone through to the monitor system. This chapter is a masterpiece of brief description, ideal for a non-technical person but still giving an accurate idea of what the equipment is and, roughly, what it can do. But when the author says that the recording console "...is best left in the hands of the engineer" you'd better believe he's not

The next chapter deals with the recording session itself — what happens when the musicians get to the studio, and goes on to explain most of the more important jargon terms (talkback, takes, slate and so on) that will be heard for the first time in the studio. It also contains one of the few sections — on paperwork — that is purely American .

The chapter on recording the basic tracks is quite short, starting with a description of how the rhythm tracks are laid down in preparation for the overdubs (separate tracks recorded after the basic rhythm tracks). Most of the chapter is (quite probably) about getting a good drum sound.

One thing the author does fail to mention, though, is that tinkering with the tuning, the damping, or using different heads may be enough to put the drummer off his timing. Drums, like all instruments, have a particular feel, different for every drummer depending on how he likes to set up his kit, and he may need some time to adjust to a different set-up.

The chapter on live recording is very clear on both the advantages and th problems of live recording. Many bands prefer to record 'live' because the method is better at capturing the magic of a band in full flight. Most rock & roll is better recorded this way, while jazz and classical music can *only* be recorded live. In rock, overdubs can still be added later using the live session as the rhythm tracks, a trick which produces (near) recording perfection while avoiding the dull lifeless quality usually created by the 'rhythm track and overdub' technique (almost always used, for example, for disco music).

Whatever method used to lay down the basic tracks, the 'rough mix' always comes next, followed by the overdub sessions, if needed (maybe the guitar player got 'hot' on take 21 and doesn't need to re-do his lead solo!).

The final mix session is just as important as all that has gone before and this chapter contains an awful lot of good advice — the author's comments on over-using flanging or echo just because its there are all too true. However the usual complement of 'outboards' — phasers, flangers, reverb chambers, noise gates and various delay lines — are all dealt with (these gadgets are called 'outboards' because they are apart from or 'outside' the mixing console, which is commonly called just 'the board').

Editing is more or less part of the final mixing process, where it may be used to lose part of a song, or to construct a usable tape out of three of four takes! Editing is a complicated business and its quite amazing what a good engineer can do with half a mile of 4in tape, while for the musician its as well to have an idea of what can — and more important, what can't be accomplished by editing.

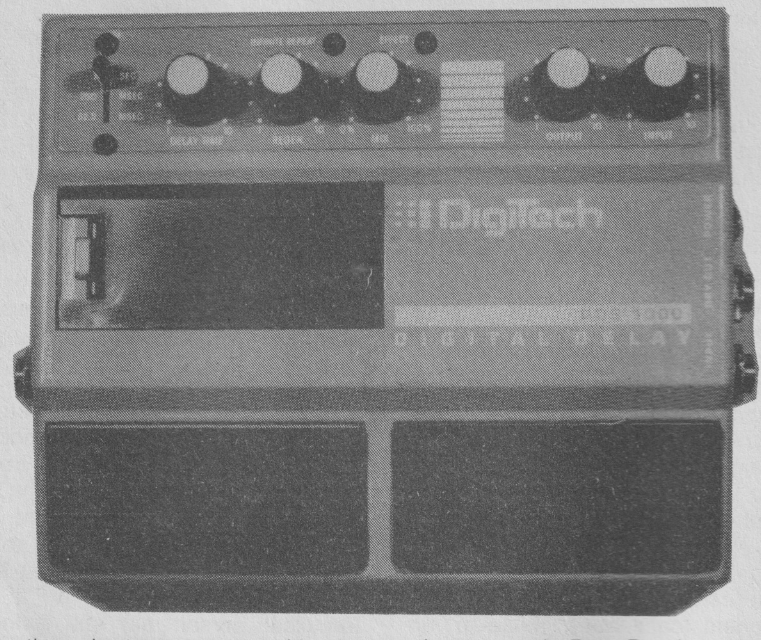
The final stage in making a record is cutting the master disc. The musician doesn't have a lot to do with this, but since it can drastically effect the sound on the vinyl, it is once more a good idea to know what can be done to rescue a dull tape, and what can be done to ruin a good one!

The final chapter in this excellent book is a glimpse into the future of recording technology — automated mixing and digital recording — and it closes with a very good glossary of terms. This will be useful as a quick reference, but in fact every term is clearly explained in the text in a way that is easy to understand and doesn't try to blind the reader with fancy science. The tone of this book as a whole makes it easy reading, while the author makes good use of his experience on 400 album sessions to cover just about every point of interest and importance in making a record.

All the books reviewed here are distributed in the UK by Book Sales Ltd., 78 Newman Street, London W1P 3LA. Good bookshops which do not stock the books should be able to order them for you. Book Sales also represents the well better-known imprint Omnibus Press.

SOUND PARTNERSHIP

Marcus Smith



SINCE I DID a feature on Boss effects pedals a couple of months ago, it has become clear that music effects are well up among the most popular projects. *Electronics Monthly* is doing a DIY pedal elsewhere in this issue, but I have been looking at some more commercial pedals of varying prices. And price is a moot point, taking into account that one top quality pedal can cost twice as much as a modest but sound guitar (***If you ignore that last comment you can call that eight times – Ed.***).

The pedals we have are by **Aria**, **Boss**, **DOD**, **Pear 1** and **T.C. Electronic**. We then asked them for two or three representative samples.

ARIA also make guitars. They are a Japanese company, but although the English is a bit strange at points, their fold-out instruction leaflets are clear. I'm not too sure about their 'one touch' battery compartment which, sure, needs only one hand to release the lever (usually), but I had a struggle to get the PP3 battery connected, and ripped a thumbnail trying to get the thing closed again. Fortunately, Aria do an AC adaptor as well.

Aria MP-5 Metal Pedal

Four controls: Distortion, Level, Low boost/cut and High boost/cut. When I first plugged this in I hated it. It might have been the torn thumbnail. With much in the way of low boost/high cut the sound is woolly and boomy; the hard distortion appears towards the end of the knob's clockwise travel and comes up suddenly, with an attendant increase in gain and feedback.

Once I started concentrating on boosting the treble I was much more at home. The sound becomes distinctly punchy and lively, sharp and sustaining. Good for many repeated choruses of *Smoke On The Water*, and I wasn't in a hurry to move on. The Low boost/cut is at 100hZ and the High boost/cut is at 10kHz, plus or minus 20dB in each case. The quoted S/N is 85dB. The thing hisses all the time, even with maximum treble cut, but this is an unavoidable side effect of high-gain distortion pedals.

Aria DD-X10 Digital Delay

Spare us! Another battery change! This time the darn thing bit my other thumb as I tried to close it. Can someone have a word with the guy who makes the cases?

Digital Delay was the latest thing in foot pedals until sampling and external triggering began to turn up. The DD-X10 has four delay ranges: 4-16msec, 16-64msec; 64-256msec and 256-1024msec. The precise delay time can be adjusted with the Delay Time control, the number of times the delay is repeated with the Feedback control and the mix of delayed and direct signal with the Delay Level control.

On the shortest range, with the Feedback and the delay time turned quite well up, you get a 'playing in a drain pipe' effect; on the next range, the notes sound metallic and cut off with a 'footstep' vibration following them

The next range gives a quick echo, from a slapback to a distinct ringing echo. With the Feedback turned up you get multiple ringing echoes. The overall sound is soft and warm rather than sharp and ringing, and the hiss is audible.

The pedal also has a latching function. In the Unlatched mode, if the pedal is depressed just after a sound is generated, the sound repeats while you keep your foot down. In the Lateched mode, if the pedal is tapped briefly just after a sound is generated, this sound will repeat itself indefinitely, You can play along to the stored riff. The memory will store just over a second. With the feedback turned up, the Hold can be used to lay one riff over another even unto several generations without losing the original entirely.

The pedal can be used in stereo, with the direct signal going from the direct output and the delayed signal from the main output

Aria CH-5 Chorus

I managed the battery change without losing any fingers here, as the spring seemed to be less fierce than the other two samples.

The pedal is very straightforward with three controls, Rate, Depth and Intensity. The rate control alters the speed of the sweep, giving a more definite vibrato as the control moves clockwise, and the depth control increases the modulation, building up to a definite wowing. Intensity balances the effect and normal levels in mono, and the stereo width in stereo.

It hisses like nobody's business, and gives an effective but unremarkable chorus.

The Aria pedals come in solid metal cases with a foot switch which I took for metal at first, but is actually moulded plastic. The knobs are domed, not recessed, but squat and tough enough to be proof against most damage; there is a rubber anti slip base on all the pedals.

BOSS make many of the most popular and innovative pedals. They span a wide price range. BOSS are made by Roland, who are best known for their synthesisers.

Boss Octave OC2

This pedal has three controls: Oct 1, which generates a note one octave below the original; Oct 2, which generates a note two octaves below the original, and the direct level, which adjusts the normal sound in respect to the effect.

Whether or not you want an octaver depends on whether you want that particular sound. It's a little like having a slightly damped bass playing in unison, and, as such, could be useful for thickening the sound or giving the illusion of a bassist if you are making your own tapes. There is said to be a world shortage of bassists.

Boss DSD-2 Digital Sampler/Delay

This is going up market with a vengeance, and is a recent arrival. It is a digital delay, with a sampler function and external triggering. All that, and you can still tread on it... Indeed, you must tread on it, to make effective use of the sampler.

The pedal has a range of delay times from 50ms to 800ms, switched by the Mode control and tweaked by the Time control.

All the effects I talked about with the Aria digital delay are here, and the sound is a bit sharper as well as being warm.

The pedal has a repeat mode called Rec/play and Play only. Sounds can be fed in, recorded and will play back whenever the pedal is pressed, and at whatever speed the pedal is pressed. The pitch of the recorded sound can be altered by turning the Time control. The recorded sound can also be played back under the control of an external rhythm machine via the trigger input

But to be fair, the pedal was not designed as a surrogate rhythm guitarist, and some points have been sacrificed to get the remaining functions into the small package. Preserving the recorded sound in memory cannot be done via the footpedal; you have to change manually from Rec/play to Play only. It takes a bit of practice to get your two or three note riff into the time allowed but once you've got it, any time you press the pedal, here it comes again.

Boss DF-2 Super Feedbacker & Distortion

The word here is 'distortion' rather than 'fuzz': it's not an over-the-top heavy metal sound, but a soft-to-middling boost from warm, reasonably punchy bass sounds to bucket-of-glass treble. It's a likeable object from the point of view of the distortion, but its special feature is the 'Feedback' function.

If a note is held, and the footpedal is then pressed and held down, the note apparently feeds back indefinitely. It is not, in fact, feeding back at all, as became well apparent when I tried a bend. The note went on all by itself. Eventually I pulled the plug on the guitar. The note went on. Only when I took my foot off the pedal did it stop.

Not only is the feedback sound sweet and convincing, but a second tone control, the Over Tone, alters the tone of the feedback note, from a full contralto to a melodic scream.

Boss pedal cases are very compact and, although they have tall, stand-up pots, the pot panel is recessed below the level of the footpedal, minimising any chance of switch damage in the case of an unplanned kick. The non slip rubber strip underneath is adequate, and their battery compartment has my definite seal of approval.

DOD are an American manufacturer. The story goes, that, if you hit a competitor's pedal with one of their pedals, the chips will fly off the other one.

DOD Envelope Filter FX25

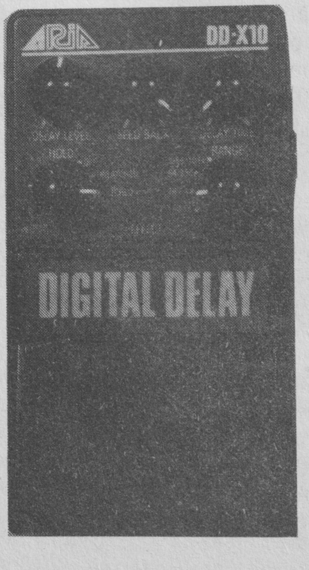
I did not have an instruction manual for this pedal — the box contained the wrong leaflet. However, as DOD's leaflets are 60% warranty listings and battery change instructions, and otherwise comprise a list of features and specifications (features being 'rubber non-skid bottom' and things of that ilk), with a small connection diagram tucked away on the back sheet, I didn't feel severely hampered.

There are two controls, Range and Sensitivity. On the single-coil, six string guitar I was using, the sensitivity had to be turned up at least half way before I got any response out of it at all.

It's a distinctive rather than a versatile effect; the sensitivity control is about right as it has to handle bass guitars and double coil pickups as well.

The Range pot alters the envelope on the note from a long drawn out wah at the lower end, to a brassy sound like a trumpet being played virogously with a mute at the upper end. The sound is almost vocal.

The volume of the direct signal drops when the pedal is engaged. This is not a case for delicate musical effects! Like all the DOD pedals, the footpedal has very little movement and engages or disengages at a touch; but this was the first pedal which made a slight, but audible, thud on the 'speaker although the list of features claims 'active switching for noiseless operation.' I couldn't get the battery panel off this one, but normally DOD battery panels are easy and quick to operate.



Each pedal has a substantial ridged rubber base — no banana skins here — and the pots are the easiest to use I have seen.

DOD Stereo Chorus FX60

The phrase 'hisses like a pit of adders' came to mind when I plugged this one in. It's a simple two control chorus (Speed and Depth). Even with both controls turned to minimum the Chorus effect is apparent and the level of the chorus is well balanced with the level of the direct signal when the pedal is disengaged

You can vary the effect from the minimum, which just gives the note a little extra ring and punch, to a sweet metallic chiming sound, through audible oscillation to a quivering, treble note at the far clockwise position. One of the attractive things about the chorus effect is that it is equally suited to chords and lead lines.

This pedal can be used for stereo, according to the connection diagram, using a line splitter to go to two separate amplifiers.

DOD Overdrive Preamp FX50

Aha! Another overdrive pedal, and up comes the noise. This one does cut out entirely when the level is turned fully anticlockwise. Disengaged, the pedal gives the guitar a fraction extra boost; with the level turned up to match the through signal, and the distortion at minimum, you still get appreciable distortion.

Through most of its range the pedal is distort rather than fuzz; towards the upper limits of the distortion, you are getting into hard, trebly fuzz which could have had more punch and definition for my taste. The pedal is very noisy and clicks ever so slightly when operated — but you are not going to notice that.

DOD also do an **'American Metal' pedal FX56**. With the distortion turned to minimum on this one, you are getting appreciable distort, and as you turn it up you get bluebottles in the loudspeaker. There is an extra control here, Presence, which varies the tone from bass towards treble. Even at a low level, raising the presence above three o'clock was a recipe for instant feedback. Punch is not known to this device — the sound is rather soft and mushy wherever you set it. It does however give appreciable extra sustain, and the level control balances the direct sound easily with the effect sound. And the case is purple.

DigiTech PDS 1000 Digital Delay

This is a fairly versatile device, although it takes a bit of time to familiarise yourself with it. The pot controls are Delay Time, Regeneration

CHECK
DIRECT LEVEL
OCT 2
OCT 1
OUTPUT
INPUT
Octave
OC-2

(called Feedback on a number of other delay pedals), and Mix, for balancing the delayed signal with the direct signal; there are also separate level pots for input and output, and a choice of three delay ranges, 15.6ms to 62.5ms, 62.5ms to 250ms and 250ms to 1000ms.

The comprehensive user booklet gives a number of different settings to experiment with: echoes, slapback, note doubling, comb filter, as well as chorus effects. It also has record and infinite repeat functions, hence the dual footpedals featured on this effect.

Up to 1000ms of delay is available for storing a riff. With the effect on, a riff can be stored if the Infinite Replay footpedal is operated as soon as it has been played. This can be switched in and out with the effect footpedal, and will stay in memory until the Infinite Replay is switched out. Sound on sound can be recorded by the same method with the Regen pot turned full up. There is no sampling facility on this unit.

The main attraction of this pedal is the clear ringing tone of the effects and the extra flexibility brought about by the dual pedals; I found it rather noisy. The case is substantial, and although the pots are tall and narrow they are far enough away from the main action not to be in the way.

PEARL are best known for their drums, but they do an extensive list of pedals as well.

Pearl FG-01 Flanger

One fact about flangers is that whether they are operating or not, while the circuit is engaged it is inclined to flange away to itself. On this one, no amount of knob-twiddling could produce a setting which was either quiet, or was not steadily flanging it's own white noise.

The instruction leaflet, ("Pearl Sound Spice Effectors — capable of meeting the most professional requirements") gave four sample settings, Flanging, Phasing, Chorus and metallic sounds such as ring modulation — all of which a flanger should be able to cope with. I duly adjusted the settings and played. But, try as I might, I could not get any punch or presence from the device at all, and the amount and variety of extra noises issuing from it rapidly became a seige. The phasing was barely audible, the chorus was thin and metallic, and when I set the feeback for the ring modulation effect, it fed back. Having plucked a discernable ring modulation from the pedal, I retreated. The pedal is still feeding back to itself, very quietly; I tried unplugging the guitar, but this makes it worse. In a minute I shall unplug the pedal. There! A phweep, and it dies.

Not discouraged enough to give up completely, I tried its sister, the **PH-03 Phaser**. Again, the device is a bit noisy and talks to itself a good deal. The leaflet gives four sample settings: Phasing, dramatic phasing, rotary speaker and surge.

The first setting gave that sweet mild muted sound with a bit of wah in the background. The effect of the sweep kills off the bass notes a bit, but this doesn't matter as the sound is best suited to melodic treble. The dramatic phasing is good for single note lines in the midrange — it could have a bit more punch but otherwise I enjoyed it, and the feedback setting showed no signs of exploding into uncontrollable feedback.

Pearl CO-04 Compressor

Compression is an effect which raises the output of the device as the original note dies away, giving longer apparent sustain. This pedal has four controls: Attack, Tone, Level and Sustain, so it is doing more than simple compression.

I set the level at around 2 to balance the direct signal with the effect signal, and tried their setting for 'long sustain on soft tones'. On a guitar which is short on natural sustain, it gave a prolonged note in the treble as well as the bass. A fair bit of distortion appeared on the bass, which I could not get rid of on any setting. A really punchy sound was not available, and clicks appeared if the string were not fingered with perfect accuracy — OK, that's the player's problem, but it's a warning!

Pearl AO-08 Analog Delay

Analog delays are cheaper than digital delays and do not give such a good quality noise-free sound, but they do many of the same tricks. This one has a Delay Time control, Feedback control, and two Level, Dry Out (direct) and Delay Out.

The usual effects — long echoes, 'crypt walking', metallic chimes, slapback and variations on the echo theme — are all there. The delay times are between 20ms and 400ms, which is a restricted range compared to a digital delay. There is no record and playback function

The Pearls are built into a stout black boxes, with low domed pots slightly recessed and easy to operate. The pots are less cramped than any of the other brands. There is a rubber anti-slip base, and the foot pedal itself is not the whole of the lower part of the front panel, as it may seem from the photos, but just the silver section. While this is neat, the danger is missing the pedal in a dark and frantic moment is increased.

T.C. Electronic are new on the effect scene. They are a Danish company and they claim that their effects are fully professional. The pedals are expensive. There is a wider-than-usual choice of effects in each unit, as is reflected in the names.

T.C. Electronic Chorus + Pitch Modulator + Flanger

The effect has four controls: Speed, Intensity, Mode and Width. The Mode switch switches between Chorus, Pitch Modulation (vibrato) and Flanger. The Speed pot is actually calibrated from 0.1Hz to 10Hz.

The sweetness and clarity of all the effects from this pedal is a joy and the range is a constant invitation to experiment. One thing you don't get on this pedal is a Feedback control, so there is no delay or echo effect.

The pedal has a number of unusual features: there is an input gain, which you can set for whatever instrument you are using for maximum signal before distortion; there is an external bypass socket for remote operation; the power LED flashes in sync with the speed setting for a quick visual indication.

The **TC XII Programmable Phaser** is similar, although this is devoted entirely to phasing. The Speed and Width pots have the same range as the Flanger; there is a choice of three filters, and the Function pot gives peak setting, a notch setting, and all points in between.

By 'Programmable' TC mean two things: firstly, you can connect two units

SPEED
DEPTH
Stereo Chorus
FX60
OUT
DOD

MANUAL
FEEDBACK
DEPTH
SPEED
FLANGER
PEARL
FG-01

together via the program jack for a synchronised stereo effect; and with the unit are provided two jack plugs with no cables connected — used with the program socket, each of these alters the pitch range of the unit, giving two extra ranges. "Phasing" conjures up visions of flowered smocks, but there is enough versatility here to give the whole concept a new lease of life.

Most of the TC pedals have a battery, in the least convenient place, which involves taking off the base with exactly the right size of Posidrive screwdriver. The Flanger however uses sufficient power that it does not have a battery option, but a moulded-on cable with a two pin plug.

T.C. Electronic Booster Line Driver & Distortion

Unlike the previous two pedals, this is not exactly noise free. There is a noise suppressor adjustment, but even this could not fully remove the background noise on this distortion pedal.

The pedal has two modes, Boost and Distortion. On Boost the pedal gives a bit of preamplification. I was getting quite a bit of hum, and the noise suppressor was in some settings fading abruptly and cutting off the note. It took a bit of trial and error to get that smooth, and I could have done with a bit more punch.

In the distortion mode the fuzzbox effect does not come into the picture till well past the halfway setting, giving plenty of choice of subtler distort effects. While there is enough fuzz to do plenty of brain damage, the pot does not veer off into the unusable/collapsing greenhouse — not at reasonable volume levels, anyway. Combining the bass and treble tone controls gives that metallic punch which I have been looking for.

T.C. Electronic Sustain + Parametric Equaliser

This pedal has four pot controls and two switches. The pots are Sustain, Gain, Function and Center (sic); the switches are Distortion, and a choice between 0.2 octave or 1 octave bandwidth. The sustain achieved is long and does not cut off suddenly of fluctuate. The two tone control pots between them cover ranges of +/-16dB and .05 to 50Hz, which is useful for most purposes when combined with the sustain. The distortion effect is very mild mannered indeed.

All the TC pedals are assembled in well-finished black diecast boxes of a standard rather than custom made type. The foot switch itself is a push button type.

TC's representative apologised for the English user's handbook not being available. The instruction sheets I did receive, however, tell you all you need to know about the devices.

Internal Rumblings

There are certain things I have been assuming about every one of these pedals, which I have not commented on individually. They all take 9V, PP3 size batteries, except in a couple of cases where the large amount of power required calls for mains operation only (this only applies to the TC pedals). They all use fully electronic switching, so that there should be no clicks when the pedals are operated; all have AC mains adaptors available, and all of them have the power switched on by insertion of the input jack

Some details: the Digitech's PCB fits neatly into the custom diecast box, and is of double sided pth (plated through hole) construction. The ICs are all in sockets, and the components are in neat rows. The new 74HCxx CMOS gates are used extensively.

Another DOD pedal examined was not quite so good. The PCB is glass fibre, but the layout is less organised. Two radial capacitors are so close together that they have to lean apart at 45 degrees. I noticed a reed relay: this will be the source of the clicks heard on one or two of these pedals.

The Aria Digital Delay pedal uses little PCBs to mount the controls on. One of these has a CMOS flip flop to switch the effect in and out. The board is socketed and tidy, and made of paper laminate. The one in my sample is bent into a shallow arc. On the track side is a small square liquorice Allsort with leads about 1mm apart out of all four sides. Presumably this is a surface mounting RAM chip.

The Pearl Flanger also has a paper laminate board and an extra PCB for switching. The effect is latched by a 4007 dual complementary pair plus inverter chip wired as a discrete clocked bistable. Don't ask me why!

The Boss Sampler/Delay also has a separate control board; the PCBs are thinner-than-usual paper laminate. The layout is neat — it has to be, because it is crammed. It uses titchy little 1/8 watt resistors and a great big 64 pin custom IC (or masked programmed microchip) with Roland's own trademark.

I was so unimpressed by the inside of TC's Programmable Phaser, which is musically one of the best. The components looked as if they had fallen into place and the wires were a bit of a rat's nest.

I decided to check a second one. The Chorus/Flanger was indeed much better put together. The PCBs are glass fibre.

Dollars And Dealers

Well: pounds, anyway. Some of the prices listed here are RRPs, some are dealer's list prices; shopping around can produce reasonable discounts.

ARIA: sample prices: DD-X10: £169; MP-5: £45; CH-5: £65. Distributors: Gigsville, Ltd., South Drive, Phoenix Way, Heston, Middx TW5 9ND.

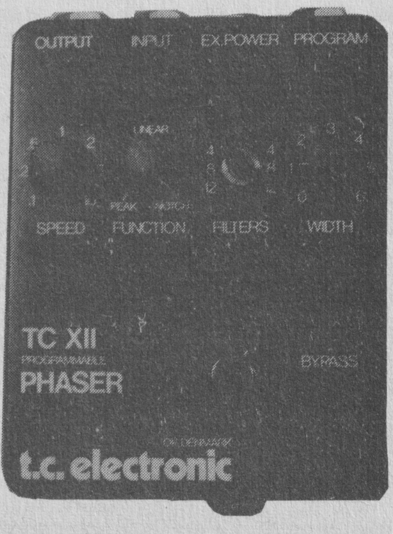
BOSS: sample prices: DSD-2: £199; DF-2: £75; OC-2 £56. Distributors: Roland UK Ltd., Great West Trading Estate, 983 Great West Road, Brentford, Middx TW8 9DN.

DOD: sample prices: FX-55: £55.95; FX-60: £89.95; FX5-56: £75.95; FX-25: £69.95; Digital PDS-1000: £199.95. Distributors: Rhino Music Spares, Burham Road, Dartford, Kent.

PEARL: sample prices: FG-01: £61; PH-03: £51; CO-04: £48; AD-08: £87. Distributor: Pearl Music Ltd., 11 Garamonde Drive, Wymbush, Milton Keynes MK8 8DF.

T.C. Electronics: sample prices: Chorus/Flanger: £157.50; Phaser: £126 (also a model in a lower octave for keyboards and bass guitars); Sustain: £94; Booster: £94.50. Distributor: Musimex, 46a Marlborough Road, London N22 4NN.

I will name the TC Phaser as my favourite out of this lot, and possibly the Aria Digital Delay second, despite its slightly lower performance compared with the more expensive models. The TCs are generally the most versatile, the Arias the best value, and Boss the best compromise between the two.



MIDI

MIDI — an interfacing system which allows electronic instruments of different types to be linked together — is now appearing on less expensive instruments. We look at how it works.

AUDIO OUT
OUT
MIDI
IN
CONTROL FOOTSWITCH

Portrait of MIDI: and input and an output socket on the back of an instrument.

A TERM which no one in the world of electronic music could fail to come across in recent times is "MIDI", which refers to a type of universal music. interface. In fact MIDI stands for "Musical Instruments Digital Interface", and although it seems to have been heralded as a new type of interface, instruments fitted with it have been available in the UK for well over two years now. The system was originally devised in 1982. However, it is only quite recently that it has been widely adopted, with a constant stream of new MIDI equipped instruments appearing on the market. MIDI interfaces tend to be associated with synthesisers, but they are now appearing on other instruments, including such things as electric pianos, drum machines, digital signal processors, and even portable keyboards.

Why MIDI?

Two reasonable questions to ask about the MIDI interface are what does it do, and who needs it. Looking at things in very simple terms, a MIDI interface enables one instrument to control another, provided they are both MIDI equipped (of course). The control could take the form of one instrument simply playing in unison with the other, or something more sophisticated could be achieved. This depends on the level of sophistication of the two instruments. What gives this type of interface particular interest is that it is not restricted to two instruments, and it is possible to connect a number of instruments of different types so that they operate as one massive system. Furthermore, the components in the system do not have to be from the same manufacturer, and any MIDI equipped instrument should be usable with any other. There is a limiting factor on the level of sophistication that can be achieved with a MIDI system, and this is due to the maximum of sixteen available channels. Sixteen note polyphony should be adequate for most users though.

The MIDI system could be of value to practically anyone who uses electronic instruments. It is perhaps of most value to modern one-man-bands, where the ability to play several instruments from one or two keyboards has obvious attractions. In general, only the more expensive electronic instruments are equipped with a MIDI interface, but it is a feature that is beginning to find its way into less expensive instruments, and which should therefore achieve wider use as time goes by.

A particularly attractive feature of the MIDI interface for many potential users is the ability to sequence a synthesiser (or any MIDI equipped instrument or collection of instruments) via this interface. The sequencing can be achieved by using a purpose build device, or a home computer fitted with a suitable interface and loaded with a MIDI sequencer program. A MIDI interface for the Sinclair ZX Spectrum computer appears elsewhere in this issue. The home computer approach has its advantages, since it gives versatility with the features of the system being largely dictated by the features of the software, and therefore being easily changed. In fact most changes would require software alterations rather than any additions to the hardware. Another advantage of using a home computer is the large amount of memory that most types provide these days. Even with a sophisticated sequencer program being run there would still be the capacity to store several thousand notes in most cases.

The MIDI interface is probably of greatest benefit to those of limited playing ability, as it enables computer composer programs to be allied with sophisticated polyphonic synthesisers. It is quite possible to have a system where (say) the user writes music onto staves on a television or monitor screen using a light-pen or whatever, and then the computer plays the music note-perfectly through a synthesiser or even a collection of instruments. Computer programs using the built-in sound generator of a computer can produce impressive results, but by using a computer and a modern synthesiser results are not that far short of having ones own orchestra. Systems of this type can be of immense help to composers and arrangers as well as those who want a complex automatic accompaniment.

There is little in common between the MIDI interface and the old system of CV and gate inputs and outputs. Information sent via the MIDI system is purely digital, and any digital to analogue conversion takes place within the instruments in the system. As many modern instruments are fully digital there is often no need for any digital to analogue conversion at all.

There are two basic types of digital

interface, parallel and serial types. With the parallel system whole bytes of data are sent at once, with each bit being transmitted over a separate line. This enables very high speed exchanges of data to be achieved, but only over very short distances (typically only a couple of metres). Longer distances give rise to problems with cross-coupling from one line to another due to stray capacitance in the connecting stable. MIDI is a form of serial interface where the data is transmitted on one signal line plus an earth line. This is relatively slow as each byte has to be sent one bit at a time, but reasonably fast data exchanges can be provided over relatively long distances. The MIDI system operates over ranges of up to 15 metres, and it is difficult to envisage a realistic situation where this would be inadequate.

Serial data transmission is much used in computer interfacing, and the MIDI system is essentially the same as the RS232C and RS423 serial systems in the way that data is transmitted, although neither of these systems is MIDI commpatible. **Figure 1** shows the way in which data is transmitted. First a start bit is transmitted, and like all the other bits it is 32us in duration. The purpose of the start bit is to indicate to the receiving equipment that it must sample the state of the signal line at 32us intervals to determine the state of each data bit, and these follow immediately after the start bit. The least significant bit is transmitted first, running through in sequence to the most significant bit, and there are eight data bits in all. Finally, the stop bit is transmitted. This bit does little more than ensure that there is a reasonable gap between the end of one byte and the beginning of the next. This is an entirely conventional form of serial data transmission, and ordinary UARTS and other serial interface devices can be used in MIDI interfaces.

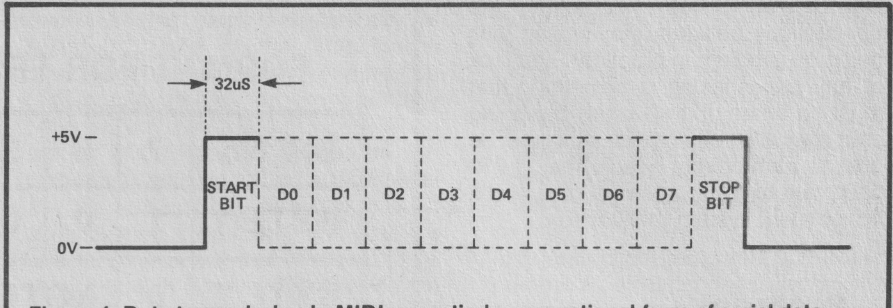


Figure 1. Data transmission in MIDI: an entirely conventional form of serial data transmission.

Baud Rate

The baud rate of the MIDI system is 31.25k baud. The baud rate is merely the rate at which bits are transmitted with an assumed continuous data flow. Originally the baud rate was to be 19.2k baud, which is a standard rate for RS232C and RS423 serial interfaces, but this was felt to be too slow and the higher rate was eventually adopted. This is in a way a pity, in that it makes it impossible to interface computer serial ports to MIDI equipment, and computer control is generally only possible via an add-on MIDI interface. Possibly MIDI interfaces will become standard on home computers though, and the new Atari 520ST is in fact equipped with one. Incidentally, although the choice of a 31.25k baud rate might seem a bit idiosyncratic, it is a practical choice in that a suitable clock signal can be generated by standard 1MHz, 2MHz and 4MHz crystals plus a binary divider (1MHz divided by 32 for instance).

Apart from the baud rate there is another difference between the MIDI system and the RS232C or RS423 systems. With MIDI systems the transmitted signal is at normal 5 volt logic levels, whereas the RS232C and RS423 systems use positive and negative voltages (typically plus and minus 12 volts in the case of the RS232C system). Furthermore, MIDI equipment uses an opto-isolator at each input to prevent problems with earth and hum loops. The standard type of connector is a 5 pin (180 degree) DIN type, with separate sockets being used for inputs and outputs. There is a third type of MIDI interface; the "MIDI Thru" type. This is a (normally) buffered output which is driven from the MIDI input, and it can be used to couple the input signal through to a third device. In fact, in theory at any rate, any number of instruments can be linked in the signal chain using the "In" and "Thru" sockets. **Figure 2** shows the usual connection arrangement. If the master device has a number of outputs it would be possible to drive each slave device from a separate output on the master unit, but this "star" method of connection is little used in practice.

Figure 3 shows the method of connection for the 5 pin DIN connector for both inputs and outputs ("Thru" sockets are wired on exactly the same manner as output sockets). Pins 1 and 3 are unused, and the earth connection at pin 2 of output sockets is only included to permit the screen on the connecting cable to be earthed. Wiring up MIDI equipment is simplied by the lack of any handshaking arrangement to provide a hold-off if a piece of equipment in the system receives data at a faster rate than it can process it. This gives only two connecting wires plus the screen (which prevents the radiation of radio frequency interference). It is merely necessary to connect pins 4 and 5 of one plug to pins 4 and 5 respectively on the other, and a standard ready-made 5 way DIN to 5

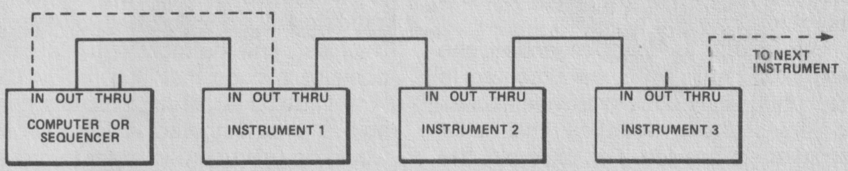


Figure 2. The method of connecting a number of instruments together using the 'In' and 'Thru' sockets.

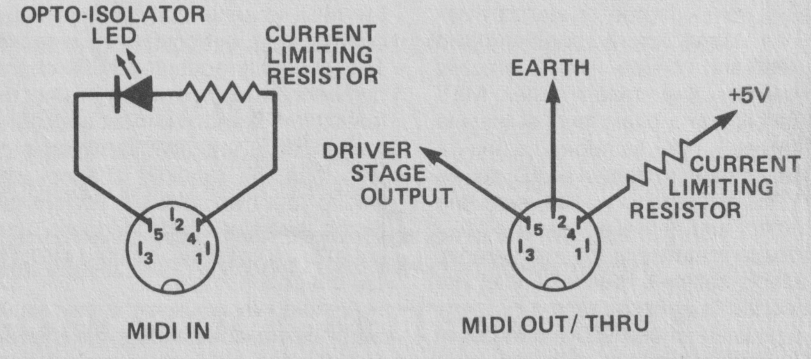


Figure 3. The method of connecting DIN connectors to MIDI. 'Thru' sockets are wired as for 'out' sockets.

way DIN audio cable will suffice (the connections between the unused pins being no problem in practice). The lack of any handshaking is something that must be borne in mind when designing your own MIDI equipment, as it must be able to comfortably handle the 31.25k baud rate to avoid a loss of data and a consequent system malfunction.

FORMAT FOR TRIGGERING A NOTE

BIT No	7	6	5	4	3	2	1	0
BYTE 1	1	0	0	1	CHANNEL No (0–15)			
BYTE 2	0	NOTE VALUE (1–127)						
BYTE 3	0	VELOCITY VALUE (1–127)						

FORMAT FOR SWITCHING OFF A NOTE

BIT No	7	6	5	4	3	2	1	0
BYTE 1	1	0	0	0	CHANNEL No (0–15)			
BYTE 2	0	NOTE VALUE (1–127)						
BYTE 3	0	VELOCITY VALUE (1–127)						

These two boxes show the data format for triggering a note, and for finishing it, as explained in the text.

Protocol

Designing a digital interface for music purposes could, with some justification, be regarded as the easy part. In order to be of real use the system must be designed so that a wide variety of instruments from various manufacturers can be used together, in any combination, and to give useful results. This produces a number of problems, one of which is simply that some instruments are inevitably going to be very much more sophisticated than others. Designing a system to suit complex instruments would preclude the use of relatively simple types in the system.

On the other hand, bringing the system down to a level that would suit even the most basic of instruments would render it of little practical value. This problem has been circumvented in the MIDI system by having several operating modes. There is a fundamental mode which is common to any MIDI equipped device, and which therefore ensures that any two MIDI devices can operate together to some degree. There are three other modes which give greater sophistication, and which allow suitable devices to be used in more sophisticated set-ups. We will consider the various operating modes in more detail shortly.

The system is an open ended one which has provision for manufacurers to have their own codes to provide access to any special features that their instruments provide. For example, the ability to change from one programmed voice to another under MIDI control is possible with some instruments, but, in general, operation at this level of sophistication is only possible using instruments from a single manufacturer. Possibly some degree of standardisation on these more sophisticated features and codes will eventually emerge. In the mean time, MIDI compatibility at a basic level at least is guaranteed. For someone using a home-computer to control a MIDI set-up there is compatibility between the computer and any advanced, MIDI accessible features of the instruments, as suitable software routines are all that are needed to access them.

It is possible to use MIDI interfaces in a fairly fundamental way with little or no understanding of how the system works, but for anyone wishing to use MIDI interfacing in a fairly advanced way it is essential to have a reasonable grasp of the way data exchanges select the required notes, gate times, etc. Bytes of data are normally sent in groups of three, and the first of these contains two types of information. The two items of information are contained in four bit "nibbles", and the most significant byte contains a code that is normally 1001 (144 in decimal), which indicates that a note is to be triggered. A different code would be used here in order to provide access to a special feature of an instrument, but as this is something that varies from one instrument to another we will not consider this aspect any further here.

The least significant nibble specifies the MIDI channel number, and is from 0000 to 1111, or from 0 to 15 in decimal. The generally accepted MIDI channel numbering runs from 1 to 16 rather than from 0 to 15, and you must accordingly remember to use a value which is one less than the required channel when writing your own software. Rationalising things, the value used to trigger a note is equal to 143 plus the required MIDI (1 to 16) channel number.

A note value is contained in the second byte, and although 8 bits give a 0 to 255 range, note values are only from 1 to 127 inclusive. This still gives an impressive range of well over ten octaves, although the full range is not achievable on every instrument. However, even if an instrument has (say) only a four octave keyboard, a greater range of notes would probably be available via the MIDI interface. This is nothing new, and with many analogue synthesisers it is possible to play pieces via the gate and CV inputs that would be impossible to produce by way of the keyboard. Middle C has a note value of 60, and each increment or decrement by one is a change of one semitone.

The third byte in the group again has a value in the range 1 to 127 inclusive, and this is a velocity value. This is only applicable to touch sensitive instruments, where a value of 1 represents minimum velocity, and 127 represents a note struck as hard as possible. To maintain compatibility between instruments that are touch sensitive and those that are not this byte must always be transmitted by a MIDI instrument, and it would normally have a value of 127 when a non-touch sensitive device is sending MIDI data. Non-touch sensitive devices are designed to simply ignore this third parameter.

Another set of three bytes are used to switch the note off again. Of course, the time from triggering a note to releasing it is equivalent to the gate pulse duration in analogue synthesiser terms, or the duration for which a key is held down in simple playing terms. The note does not necessarily switch off instantly — this depends on the envelope shape that has been selected. The first byte differs only in that the most significant nibble becomes 1000 (128) in decimal. The first byte therefore has a decimal value which is equal to 127 plus the MIDI channel number. The second byte is the note value, as before, and the third byte is the velocity value (which is normally 0 when a non-touch sensitive device is transmitting data).

MIDI modes

There are four MIDI modes, although only three of these are in general use. Originally these modes were called "omni", "poly", and "mono", but these days they are often just given identification numbers (1, 3 and 4 respectively). Mode 2 (which has no name as far as I know) is one that is only featured in Yamaha synthesisers, and it has every channel of every instrument in the system play the same notes. As this mode is not generally applicable, and is not one that I have ever encountered, it will not be described in more detail here.

Mode 1 ("omni") is the most important one in that all MIDI equipped instruments must have this one (the others are optional), and they must switch to this mode automatically at switch-on. This is the most basic operating mode which ensures a reasonable degree of compatibility between any two MIDI devices. When set to this mode an instrument will respond to note on/off data regardless of its channel number. With sophisticated instruments it is possible to select the channel or channels to be used, but with simple MIDI devices the channelling may be preset. This mode ensures that two devices can exchange data even if they lack channel compatibility. Just how the voices of a polyphonic instrument respond to the received data depends on the internal assignment concerned. Normally the received data will be treated as if it originated from the keyboard, and with most polyphonic instruments this results in the first note being played on voice 1, the next on voice 2, and so on. Whatever the method of internal assignment, a polyphonic instrument is effectively reduced to a monophonic type if it is used as the slave instrument.

In mode 3 ("poly") it is possible to assign instruments to a particular channel, so that data can be sent to just one instrument in the system. Of course, if desired it is possible to assign two or more instruments to the same channel so that they operate in unison, but independently of other instruments assigned to different channels. As for mode 1, the way in which an instrument responds to the received data depends on its method of internal assignment.

The most sophisticated of the three main modes is mode 4. This enables each voice of a polyphonic instrument to be given its own MIDI channel, with the only restriction being that the channel number must run consecutively. This is of great value if you wish to sequence a polyphonic instrument from a computer (or a purpose built sequencer), as it permits polyphonic multi-voice music to be produced.

Conclusion

Even using a MIDI system which just consists of a single polyphonic synthesiser plus a computer or custom built sequencer, the capabilities of the system are tremendous. Things are certainly must neater and conveninent than an equivalent analogue system using separate gate and CV sockets for each channel, and MIDI gives the advantage of encompassing touch sensitivity. It also leaves room for tremendous future expansion, and could easily accommodate virtually any feature imaginable.

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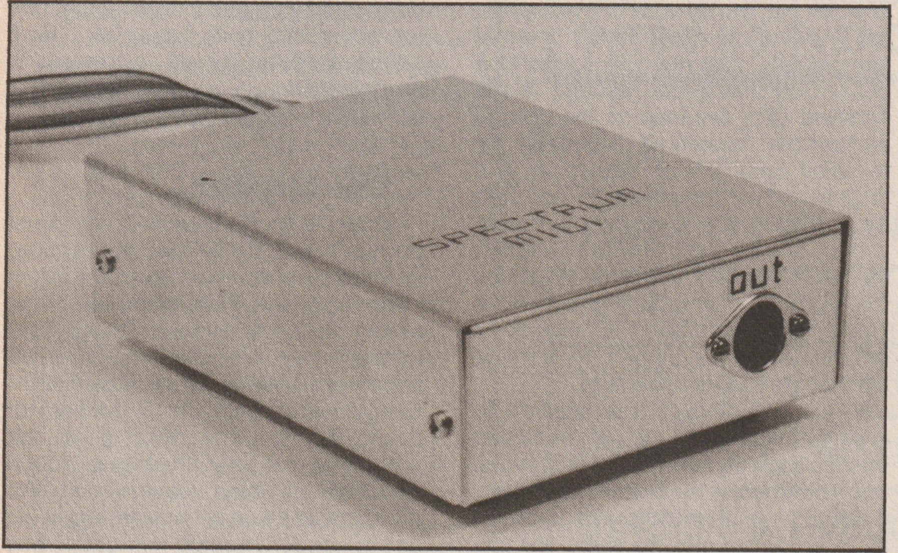
SINCLAIR SPECTRUM MIDI INTERFACE



A MIDI step time sequencer using a ZX Spectrum, which turned out to be a good-value memory source.

R.A. Penfold

THE WHYS and the wherefores of MIDI interfacing are covered elsewhere in this issue. This complementary article describes a simple interface which enables a Sinclair Spectrum or Spectrum Plus computer to operate as a step-time sequencer for any MIDI equipped instrument or set of instruments. The Spectrum is an obvious choice for this application since it provides a large amount of memory for a very modest outlay. The 48k versions are preferable for an application of this type, but even the 16k version enables very long sequences of notes to be programmed (more than many purpose built sequencers, in fact). Even if the Spectrum computer had to be bought especially for use as a sequencer (and there must be a few people who do not have access to one), together with this interface it would still be an economically viable proposition and would in fact represent excellent value.

It will be assumed here that the reader has a basic understanding of the way in which the MIDI system functions. Anyone who is unfamiliar with MIDI interfacing but who is interested in building this project should therefore refer to the MIDI feature.

UART

The MIDI type of interface is just a simple asynchronous serial type using a standard 1 start bit, 8 data bits, 12 stop bit, and no parity word format, with a baud rate of 31.25k. There are several devices which could be used as the basis of a MIDI output interface for the Spectrum, but the choice basically boils down to the few serial interface chips specifically designed for systems based on the Z80 microprocessor (which does, of course, include the Spectrum), or UARTs (universal asynchronous receiver — transmitters). UARTs will interface readily to most logic circuits, whether microprocessor based or not, and will certainly operate with the Spectrum without any difficulties. The device finally selected for this interface is the industry standard UART, the 6402. This is generally somewhat cheaper than the Z80 serial interface chips, is equally convenient from the hardware point of view, and is slightly easier to use as far as the driving software is concerned.

System Operation

The block diagram of **Figure 1** helps to explain the way in which the interface functions. The data to be transmitted in serial form is provided by the computer as bytes of parallel data on the data bus. There is a constant stream of data on this bus, and some means of selecting only the bytes intended for transmission is needed. It is achieved by decoding some lines of the address and control buses, which take up a certain combination of logic states only when data is written to the interface. The address decoder detects the appropriate combination on the monitored lines, and activates the buffer stage. The latter is actually an 8 bit data latch, and it transfers each byte of latched data to the transmission register.

This register is a form of shift register, and it shifts out each bit of the byte, starting with the least significant bit (D0) and working in sequence through to the most significant bit (D7). In practice things are a little more complex than this, as the stop and start bits must be added. However, this is all taken care of via the control logic block which is programmed to give the required word format by wiring input pins of the UART to the appropriate logic levels. This is the main

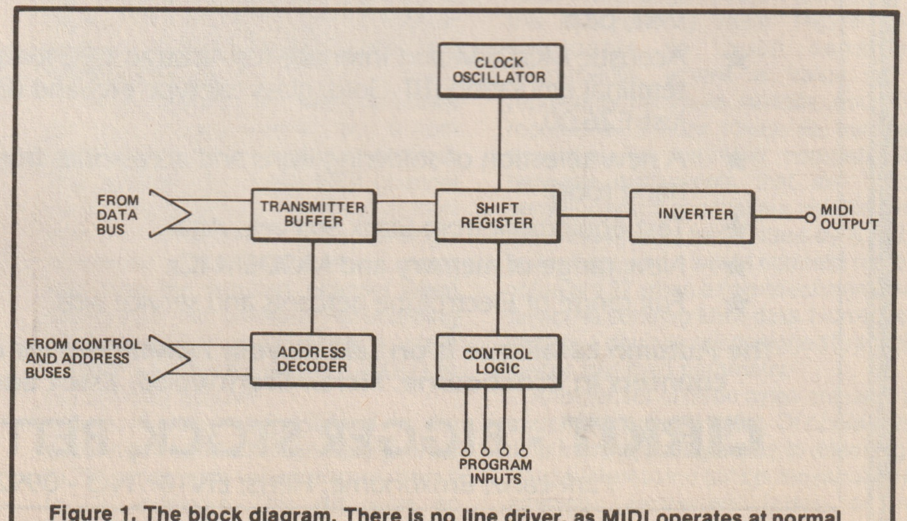


Figure 1. The block diagram. There is no line driver, as MIDI operates at normal logic levels.

respect in which UARTs differ from serial interface devices which are designed specifically for use with microprocessors, where the word format is under software control. As only one word format is required in this application, the ability to change the format under software control is of no value and a UART is actually a little more convenient in use as there is no need for any setting up routine to set the required format.

The output from the UART has the wrong polarity to drive a MIDI input, and an inverter stage is used to process the signal to give a signal of the right polarity. Normally a level shifting 'line driver' circuit would be used here to give the plus and minus 12 volt signal levels used in the RS232C serial system. However, the MIDI system operates at normal logic levels, and an ordinary logic inverter takes the place of a line driver.

A clock oscillator is needed in order to set the desired baud rate. A crystal oscillator ensures adequate stability and avoids the need for any setting up. The UART has a built-in divide by 16 circuit at the clock input, and a 500kHz clock frequency is therefore needed (31250Hz × 16 = 500000Hz or 500kHz).

Circuit Operation

The full circuit diagram of the interface appears in **Figure 2**, and the use of a special asynchronous interface device enables an extremely simple circuit to be produced.

The simple method of input/output addressing adopted for the Spectrum computer enables a very basic address decoder to be used. The normal system of Z80 input/output mapping has only the eight least significant address lines decoded, giving 256 addresses from 0 to 256. The Z80 has separate input/output and memory maps, with IORQ being set low when the microprocessor wishes to access an input/output circuit, and MEMRQ being set low when it wishes to access a memory device. The Spectrum's system of addressing differs only in that all sixteen address lines are used for input/output addresses, giving an address range of 0 to 65535. On the face of it this is a more complex method with twice as many address lines to be decoded. In practice only partial address decoding is used, with the basic system being to take a particular address line low to activate each input/output circuit.

In its most simple form address decoding just consists of monitoring IORQ and one address line, with the input/output circuit being activated when both lines go low. A5 is an address line which is available for user add-ons, and this output going low (with all the other address lines high) corresponds to address 65503. Note that this is address 65503 of the input/output map and not of the memory map. Accordingly, data is written to or read from this address using the BASIC OUT instruction and IN function, not POKE and PEEK respectively.

Here the address decoding is slightly more complicated in that the RD (read) and WR (write) lines of the control bus are also decoded. RD goes low when the microprocessor reads data from memory or an input/output device, whereas WR goes low during write operations. Although it would seem that we only need to write to the UART in this case, it is in fact necessary to read one of its status outputs as well when using a machine code sequencer program. This ensures that data is not written to the UART faster than it can transmit it.

IORQ and A5 are taken to negative enable inputs of IC2, while RD and WR are taken to address inputs 0 to 1 respectively. Address input 2 is not required and is simply wired to ground. This results in output 1 pulsing low when data written to address 65503, and output 2 pulsing low when data is read from this address.

IC1 is the UART, and its eight data inputs are connected to the corresponding lines of the data bus. The 'write' output of IC2 (pin 14) drives the TBRL (transmitter buffer register load) input of IC1, and latches data into IC1 when data sent to the correct address. The TBRE (transmitter buffer register empty) output is connected to line D7 of the data bus, and this three-state output is brought to the active state when the 'read' output of IC2 goes low. In other words this line can be read by reading bit 7 at input/output address 65503. TBRE goes high when IC1 is ready to receive data, and low while the transmitter buffer is full. A software routine can be used to provide a hold-off until TBRE goes high, thus preventing characters being written to the full register and data being lost.

IC1 requires a high reset pulse at switch-on, and this is provided by the simple C – R charge circuit, C1 – R1. Q3 is a straightforward common emitter switching transistor which acts as the inverter at the output of IC1. R8 provides current limiting at the output. The 500kHz clock signal is generated by Q2 and its associated components, which form a well known crystal oscillator configuration. X1 is the crystal, but in practice it is perfectly acceptable to use a 500kHz ceramic resonator here. This

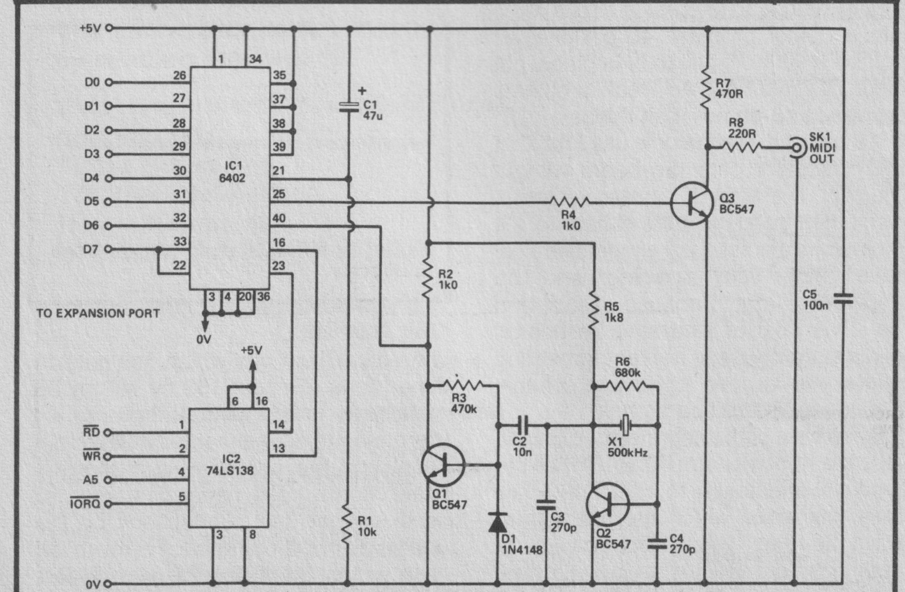


Figure 2. The circuit. Note that IC1 is a CMOS type, and the interface draws its power from the Spectrum.

Parts list

RESISTORS

(All ¼W 5% carbon)

R1	10k
R2,4	1k
R3	470k
R5	1k8
R6	680k
R7	470R
R8	220R

CAPACITORS

C1	47uF 10V radial elect
C2	10nF polyester
C3,4	270pF ceramic plate
C5	100nF ceramic

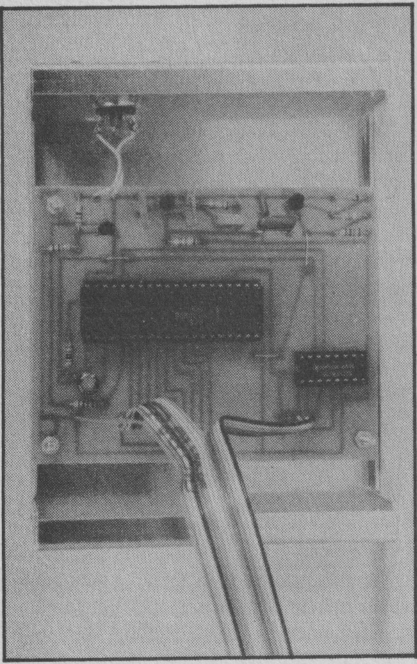
SEMICONDUCTORS

IC1	6402
IC2	74LS138
Q1,2,3	BC547
D1	1N4148

MISCELLANEOUS

X1	500kHz ceramic resonator or crystal
SK1	5 pin DIN (180 degree) socket

2 × 28 way 0.1 inch pitch edge connectors with polarising key; case about 133 × 102 × 38mm; 14 way ribbon cable; PCB, wire, solder, etc.



has the advantage of lower cost, and although it provides inferior accuracy and stability than an ordinary crystal, performance is still considerably more than adequate in both respects. Q1 amplifies and clips the output of Q2 to provide a suitable signal to drive the clock input of IC1 properly. Although the 31.25k baud rate required for a MIDI interface is quite high by serial interface standards, the 6402 is being used comfortably within its maximum clock frequency rating of 4MHz.

The 6402 UART can handle a wide range of word formats, including 5, 6, 7 and 8 data bits, one or two stop bits, plus odd, even, or no parity. In fact some twenty four different formats are available, including the 8 data bits, 1 stop bit, and no parity required in this application. Five inputs are tied to the required logic states in order to select the required format, and these are pins 35, 37, 38 and 39 (which are tied high), and 36 (which is tied low).

IC1 does, of course, include a serial receiver circuit, but for a step-time sequencer only a MIDI output is needed, and the receiver section of IC1 is therefore left totally unused.

The interface requires only a single 5 volt power supply, and the Spectrum's expansion port provides a suitable output. Although serial interface devices are fairly complex and normally have quite high current consumptions, the 6402 is a CMOS type which has a very low current consumption (about 2 milliamps in this circuit), and the +5 volt output of the Spectrum can comfortably supply the total current requirement of the interface.

Construction

All the components (apart from output socket SK1) fit onto the printed circuit board, as detailed in **Figure 3**. IC1 is a fairly expensive device, and it is a CMOS type. It is consequently essential to rigidly observe the basic antistatic handling precautions. Fit the device in a 40 pin DIL IC holder, but do not plug it into the holder until all the other components have been soldered in place and all the wiring has been completed. IC1 should be supplied in some form of protective packaging (usually a plastic tube or conductive foam), and it should be left in this until it is time for it to be fitted in place. When IC1 is plugged into place try to handle the pins of the device as little as possible, and be careful not to buckle any of the pins under the device. It is often necessary to bend the pins inwards slightly before an integrated circuit will fit into a holder properly. 40 pin devices can be a little difficult to fit in place, but there shoiuld be no real problems provided you do not rush things.

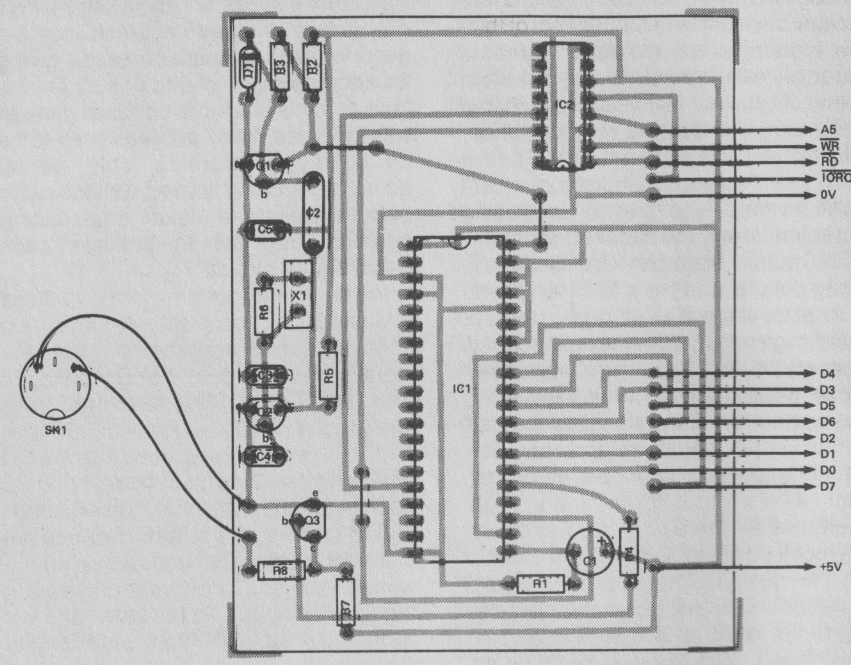


Figure 3. The PCB overlay. As IC1 is CMOS, and not cheap either, so handle it carefully. An IC socket is an absolute necessity.

If a ceramic resonator is used for X1 it will fit straight onto the board without difficulty. If a 500kHz crystal is used it might then be a little more difficult to fit it in place as a crystal will almost certainly have wider pin spacing and be physically larger. Because of this, and the lower cost of a ceramic resonator, the use of a crystal is not recommended unless you happen to have a suitable component in the spares box.

Make sure both integrated circuits are fitted the right way around as IC1 has the opposite orientation to IC2. There are three link wires which are made from about 22 swg tinned copper wire, or trimmings from resistor leadout wires will suffice. Pins are fitted to the board at the two points where connections to SK1 will be made.

The board is connected to the expansion port of the Spectrum via a 14 way ribbon cable about half a metre or so in length and terminated in a 2 by 28 way 0.1 inch pitch edge connector. The Spectrum has provision for a polarising key, and connectors fitted with a suitable key are readily available. Wiring details for the edge connector are shown in **Figure 4**, and the connections on the printed circuit board are grouped in a way which matches up well with edge connector. However, it is still essential to take great care to avoid errors when wiring up the connector, and to thoroughly check everything once you have finished.

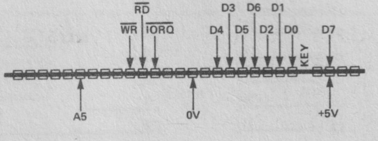
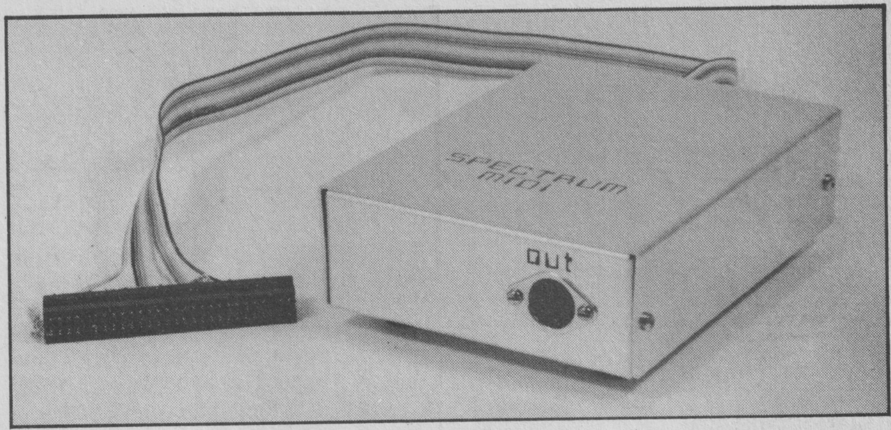


Figure 4. Wiring details for the edge connector.

An aluminium box which has outside dimensions of about 133 by 102 by 38 millimetres makes a good housing for this project. SK1 is mounted on the front panel and the printed circuit board is fitted on the base panel. Spacers are used to keep the connections on the underside of the board clear of the metal case so that short circuits are avoided. The two connections from the board to SK1 are then added, being careful not to accidentally cross over the leads. It might be necessary to cut an exit hole for the computer lead in the rear panel of the case, but provided a thin ribbon cable is used there will probably be no difficulty in simply taking the cable through the small gap between the two sections of the case.



In Use

It is important to connect the unit to the computer before the computer is switched on. Once it has been switched on, the usual initial screen display should be obtained after the brief setting up period. If there is any sign of abnormal operation you should switching off at once and recheck all the wiring.

If all is well, the unit is ready for connection to the synthesiser (or whatever) and testing. A simple monophonic sequencer program is provided in the accompanying listing, and should prove useful for testing purposes and familiarisation with the MIDI system. Much greater sophistication is possible though, should you require it, and there is plenty of scope for software writers to produce complex sequencer programs including such things as graphics displays. When using BASIC the unit is very easy to use, with the required data being written to address 65503 using the OUT instruction. The interface can output data faster than the computer can supply it, making it unnecessary to monitor the status output.

One important point to bear in mind if you wish to have multichannel sequencing, is that BASIC is not really fast enough to give good results with notes playing in close syncronisation if more than about three channels are operating. This really requires the speed of a machine code program. When using machine code remember to monitor bit 7 at address 65503 to provide a hold-off until the transmitter buffer register is empty (bit 7 goes high). Alternatively, a delay loop could be used to provide a hold-off of about 35us in duration between each byte of data being written to the interface.

Shop

The components for this project are widely available. Most can be obtained from Maplin (0702 554155), including the 6402 UART, which costs £7.74. The ceramic resonator is supplied by Cirkit (0992 444111).

The components should cost around £15 excluding the case and PCB. The PCB will become available from our PCB service in due course.

Careful shopping around may produce cheaper prices on the expensive components, but we cannot advise on component substitutions.

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ELECTRONICS FROM THE START

Operational Amplifiers don't just handle DC: they can be connected to an AC voltage source. But what is an AC voltage?

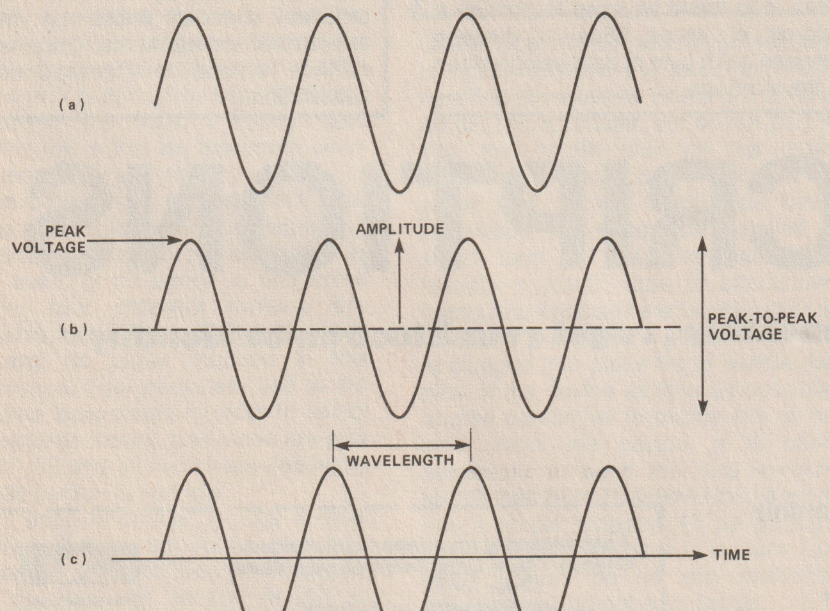


Figure 1. The most commonly used AC voltage is a sine wave. Amplitude, peak-to-peak voltage and wavelength are illustrated here. Frequency describes the number of wave-cycles per second.

Keith Brindley

The op-amp can be used in two basic circuits: non-inverting (where the output voltage is the same polarity as the input voltage); and inverting (where the output voltage is the opposite polarity).

These two circuits were used merely to amplify DC voltages — in fact, the op-amp is often called a DC amplifier. But this doesn't mean to say that an op-amp can't be used to amplify AC voltages. An alternative way of considering it is to think that DC doesn't necessarily stand for *direct current* but that it stands for *direct connection*. Because this is really all the term DC means with respect to an op-amp. If you think of the circuits we saw last month, you'll remember (won't you?) that connections between the inputs and outputs of the circuits were made directly ie, no other components such as capacitors were used between the circuits and the voltage source — direct connection, geddit?

We can therefore directly connect an AC voltage source (instead of the DC voltages sources — batteries — we used before) to the circuits and they will amplify them to produce larger AC voltages. But what is an AC voltage? Well, the AC voltage which is most commonly used when designing, building and testing electronic circuits is shown in **Figure 1a**. It is called a **sinewave**. Any AC voltage such as a sinewave has two important attributes, illustrated in **Figure 1b & c**.

First, it has a particular size, called its **amplitude**, which is merely the voltage between its middle, or common voltage (usually 0V) and its highest or **peak** voltage (**Figure 1b**). Sometimes, its voltage may be given as the difference between the two peaks (at either side of the common voltage). In this case the size is known as the **peak-to-peak voltage** (abbreviated to pk-pk, or p-p voltage), and quite obviously is twice the amplitude.

Second, it has a particular rate of repetition, known as its **frequency**, which is merely the number of times it repeats itself in a given period of time. Frequency is generally measured in cycles per second, where a cycle is one complete **wavelength** of the sinewave, and a cycle per second is commonly known as **Hertz** (abbreviated to Hz).

Representing Sounds

AC waveform, which we are most likely to be interested in, are waveforms which can represent the sounds we can hear (between about 100Hz and 20kHz ie, 20,000Hz). In more complex electronics, engineers are interested in those frequencies which occur in, say, radio circuits (50kHz to about 300mHz ie, 300 megahertz or million Hertz), television (300MHz to 3000MHz) radar (3000MHz to 300,000MHz). But the principles we'll learn here in *Electronics From The Start* are the same principles which those higher frequency areas of electronics rely on.

Sinewaves aren't the only type of AC waveform: in fact we use them only because they are a standard wave. Other kinds of waves are squarewaves and triangular waves, shown in **Figure 2**. Also shown are some waveforms which are more likely to occur as sound waves — they're not necessarily sine, square, or triangular, but shapes all of their own. Nevertheless, they share the same attributes as sinewaves and square waves and triangular waves — they are said to be **periodic** ie, they repeat at regular intervals, and so are measured in frequency as Hertz.

Figure 3 shows an example of a non-inverting amplifier with a gain of two, being used to amplify an AC voltage. The input AC voltage signal (a sinewave) is amplified, so that the output voltage is twice its size. The amplifier does not

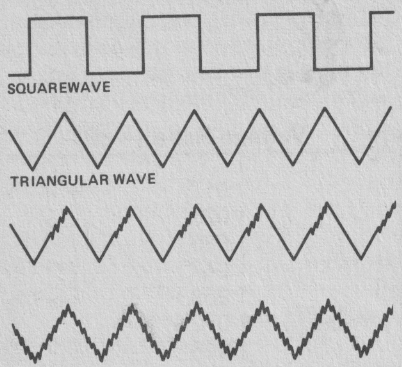


Figure 2. Other AC waveforms: squarewaves, triangular waves and other periodic waveforms.

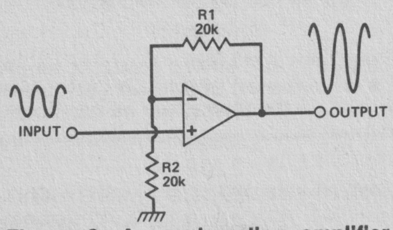


Figure 3. A non-inverting amplifier circuit with a gain of 2, amplifying a sine wave.

affect (ideally) any other property of the applied signal.

Of course, you would expect this of a non-inverting amplifier, wouldn't you? But what about an inverting op-amp amplifier? How does that affect the applied signal? Well, **Figure 4** shows an inverting amplifier with a gain or two, together with an applied AC signal. The output signal is, as you should expect, twice the amplitude of the input signal, and it is inverted, so that any point on the input signal can be seen to be the opposite polarity on the output signal. This creates the illusion that the output signal is 'upside-down' with respect to the input voltage.

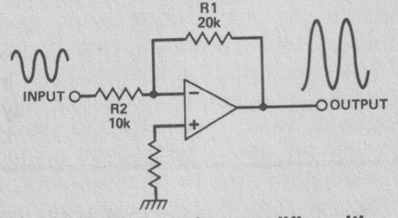


Figure 4. An inverting amplifier with a gain of 2: the output is of opposite polarity to the input.

If you know anything about maths (*Or if you read* Maths Without Misery *last month — Ed.*), you'll understand that a single wavelength of a sinewave (shown in **Figure 5**) is represented by 360°. So the inverting effect of the inverting amplifier can be represented mathematically by considering that the input signal is merely **shifted** in phase by 180° (**Figure 6**). For this reason the output signal of an inverting amplifier is often said to be **phase shifted by 180°** or sometimes **180° out of phase**. This, incidentally, is the case whenever two equally shaped waves with the same frequency are 'upside-down' with respect to one another.

AC Gain

Earlier on, it was said that an op-amp does not affect other properties of the applied input signal. So a sinewave, or a squarewave, or any wave which is applied to an op-amp amplifier is merely amplified by the amount given by the op-amp's gain calculation. In ideal life this would be true, however, things (yes, you guessed it) are not ideal.

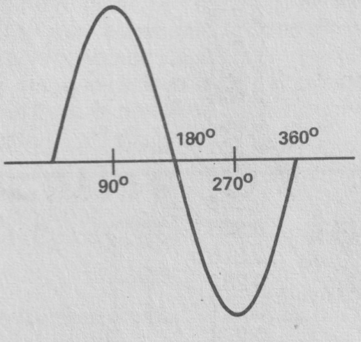


Figure 5. Like a circle, the sinewave cycle is represented by 360 degrees of movement.

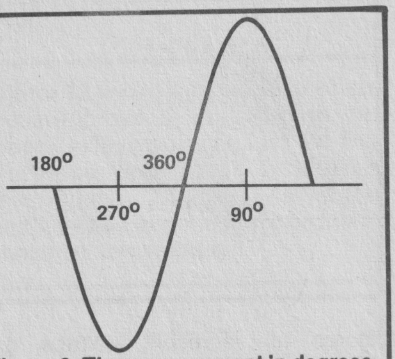


Figure 6. The measurement in degrees is always taken from the same point on the cycle. A different stage of the waveform's travel is 'phase shifted', in this case by 180 degrees.

Last month we saw that an op-amp has an extremely high gain. A 741, for example, has a typical open-loop gain of 200,000. But this open-loop gain is *at zero frequency* — look back to last month's issue of *Electronics Monthly*, and you'll see that we said just that, then. Zero frequency can be thought of as a periodic waveform with no frequency ie, it doesn't vary with time ie, it is a fixed voltage ie, it is DC. But the waveforms we are looking at here are AC. So what is the op-amp's open-loop gain at frequencies other than zero?

Figure 7 shows a graph of a 741's typical open-loop gain, plotted against frequency. You can see that below about 5Hz the open-loop gain is the same ie, 200,000. Above this, however, the open-loop gain falls dramatically. For a 741, the gain falls to zero at about 1MHz, but other types of op-amps may do this at lower or higher frequencies.

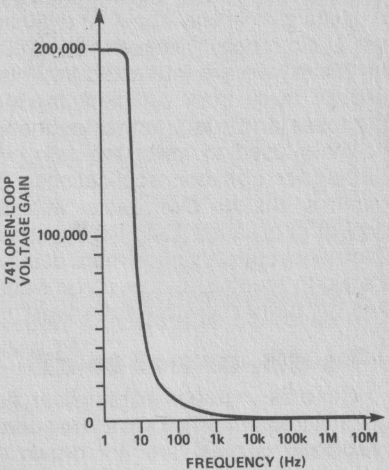


Figure 7. A graph of op amp voltage gain plotted against frequency...

This has some drastic implications for op-amp circuits. If the op-amp is used in open-loop gain mode (ie, with no feedback) then an input signal which consists of many different frequencies (such as speech or music) would not be amplified equally — higher frequencies would not be amplified as much as lower frequencies. Using feedback, we can however reduce the amount the lower frequencies are amplified. Looking at the graph in **Figure 7**, what do you think would be the highest gain of an amplifier which is to amplify all signals equally upto 10kHz? Well, by drawing a vertical line up from the 10kHz mark on the horizontal axis (**Figure 8**), we can see that the highest gain can only be very low — let's say 100. And if the amplifier were to amplify frequencies upto 100kHz, then it can only have a gain of much less — say 10. So, the higher the frequency we wish to amplify, the lower the gain which is possible. You can see

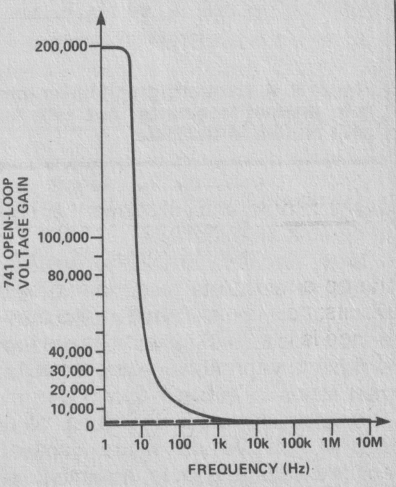


Figure 8...and the highest gain available at 10kHz: not a lot.

that as far as a 741 is concerned, it gives a useful gain of only about 40 or 50 over the audio range of frequencies (100Hz to 20kHz) we are interested in. This is usually more than sufficient for most purposes, and in fact, further feedback is normally used to lower the gain even further, for common applications. This month's *Inside Out* deals with this aspect of op-amps (and the op-amps's internal circuits) in much more detail, so it's worth reading, it, now, if you want to find out further how op-amps work.

To dB, or not to dB

Generally, graphs of amplifier gain against frequency (known as **frequency response curves**) are not drawn the way we have shown in **Figure 7**, by measuring the vertical axis in direct units of gain, but are drawn by measuring the axis in **decibels**. Now decibels (shortened to dB) are simply just a way of representing a ratio (in this case gain). They are calculated by the expression:

$$20 \log_{10} V_0/Vi$$

which gives the result of the gain in decibels.

The advantage of using decibels to show gain is that decibels are a logarithmic measure, and because the effects of gain vary in a logarithmic way the use of a logarithmic measure on the vertical axis means that linear steps are produced. This allows the frequency response curve of **Figure 7** to be transmogrified into the straight-line curve of **Figure 9**. Such a curve is much easier to draw and yet shows the results in no less an exact way.

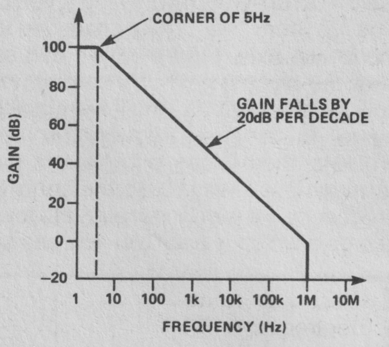


Figure 9. A standard graph of op amp gain against frequency, but with the gain plotted in decibels.

Now, decibels aren't the easiest of things to calculate (yes, you know the formula, but you still need a calculator or a set of log tables) so we've reproduced a fairly comprehensive list of voltage gain ratios in **Table 1**, along with their corresponding decibel values. Notice that a positive dB figure implies a positive voltage gain ie, an actual gain; but a negative dB figure implies a negative gain ie, an attenuation!

Table 1
Decibels and voltage ratios

Voltage ratio (-dB)	Decibel figure	Voltage Ratio (+ dB)
1	0	1
0.989	0.1	1.012
0.977	0.2	1.023
0.966	0.3	1.035
0.955	0.4	1.047
0.944	0.5	1.059
0.933	0.6	1.072
0.912	0.8	1.096
0.891	1	1.122
0.841	1.5	1.189
0.794	2	1.259
0.75	2.5	1.334
0.708	3	1.413
0.668	3.5	1.496
0.631	4	1.585
0.596	4.5	1.679
0.562	5	1.778
0.501	6	1.995
0.447	7	2.239
0.398	8	2.512
0.355	9	2.818
0.316	10	3.162
0.282	11	3.55
0.251	12	3.98
0.224	13	4.47
0.2	14	5
0.178	15	5.62
0.159	16	6.31
0.126	18	7.94
0.1	20	10
3.16×10^{-2}	30	3.16×10
10^{-2}	40	10^{2}
3.16×10^{-3}	50	3.16×10^{2}
10^{-3}	60	10^{3}
3.16×10^{-4}	70	3.16×10^{3}
10^{-4}	80	10^{4}
3.16×10^{-5}	90	3.16×10^{4}
10^{-5}	100	10^{5}

One thing you have to remember when using this table is that these figures all represent *voltage* gain. Another kind of gain — power gain — is often measured with respect to amplifiers, but this displays *different decibel values*. We'll be looking more closely at decibels; voltage, power, and others, in a later issue so keep reading.

All Bodes Well

Straight line curves such as these are known after their inventor as **Bode plots**. They are useful (apart from being easy to draw) in that they show certain characteristics of the frequency response in a better light. For instance, in **Figure 9**, the straight line of the curve above the **corner frequency** at about 5Hz falls at a rate of exactly 20dB per decade. That is, every time the frequency increases by a factor of 10 (a decade) the gain falls by 20dB (a factor of 10). Other Bode plots may decrease at a greater or lesser rate, and may indeed increase and not decrease — different circuits produce different shaped plots.

Way back in the April issue we saw some other straight line graphs, too — of simple capacitor-resistor filters. From those you should be able to work out that an op-amp is really only a type of low-pass filter ie, it allows low frequency signals to pass, while not allowing high frequency signals to pass so well. The difference between a capacitor-resistor low-pass filter and an op-amp low-pass filter is that the op-amp also has gain, so that most frequencies of signal are amplified, too. A capacitor-resistor filter only consists of two **passive** components (ie, they cannot amplify the signals) so they form what is known as a **passive filter**. An op-amp, on the other hand, can amplify the signals in an **active** way, so it forms what is known as an **active filter**— it can create a gain in signal size.

By using feedback such as we saw last month, the open-loop frequency response of an op-amp may be adjusted so that the circuit's closed-loop frequency response is clearly defined. Say, we require a circuit with a gain of 10

(20dB), then we could draw a corresponding Bode plot (**Figure 10**) which defines the frequency response. From this, we can see that the corner frequency of the amplifier occurs at about 60kHz.

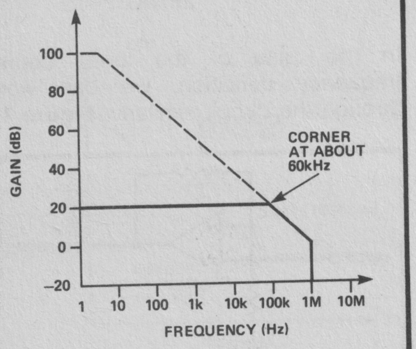


Figure 10. Using a Bode Plot ('straight line curve') the corner frequency of a circuit can be easily defined.

Now, in many cases this may be quite adequate. But let's say that we only require the amplifier to amplify signals up to about 20kHz (it's for audio frequency signals). If we used the amplifier which produced the frequency response shown in **Figure 10**, then any high pitched whistles, hisses, or other extraneous noises produced by the system within the range of about 20kHz to 60kHz will be amplified along with the wanted sounds. Similarly any low frequency bumps, rumbles or other noises below about 100Hz will also be amplified. You could argue that because these noises are outside of ordinary hearing range it doesn't matter that they are amplified together with those sounds within hearing range, but in fact it does matter, because they might interfere with the system itself. Take the case of an audio hifi amplifier, with loudspeakers attached, playing some music. If low and high frequency noises outside of the hearing range were amplified along with wanted sounds then the loudspeakers themselves may well be overlooked and damaged by the noises — even though we can't hear them! The next thing you know is: PHUT! and the loudspeakers are deaded.

So what can we do to counteract this? How do we define the frequency response of the system to be exactly what we want? Well, the answer's quite simple, we merely add components to the amplifier to artificially do the job for us. Remember we said ealier that the open-loop frequency response was similar to that of a capacitor-resistor low-pass filter. Well, we can add capacitor-resistor circuits to the op-amp, with responses similar to those which we want, so that they will combine with the op-amp's response to give the overall response which we want. The capacitor-resistor filters give the corner frequencies which we desire, while the op-amp gives the gain which we need. In effect, what we will produce is a controlled active filter.

We'll do the task in two steps. First we'll design the amplifier (say a non-inverting amplifier) to have an upper corner frequency of 20kHz. Then we'll do the same, producing a lower corner frequency of 100Hz. Let's start with the basic non-inverting amplifier we know and love, in **Figure 11**. We need to define the basic voltage gain of the amplifier, which for want of anything better to do we'll say is to be five (14dB).

From the expression for gain of a non-inverting op-amp amplifier:

$$G = R1/R2 + 1$$

We can see that the ratio of resistor R1 to resister R2 must be four to give the required gain, so we can choose some arbitrary resistance values (say, resistor R1 is 47k, and resistor R2 is 12k, which are the nearest perferred values to give this ratio).

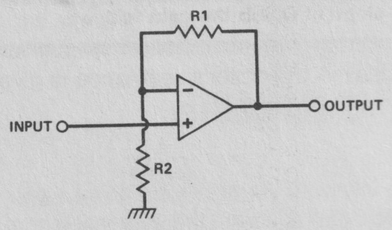


Figure 11. Our non-inverting amplifier circuit...

Figure 12 shows the circuit, to which a capacitor has been added in the feedback loop, which can fulfil the first part of the problem by defining the upper corner frequency at 20kHz. As yet the capacitor value is not shown, because we have yet to calculate it.

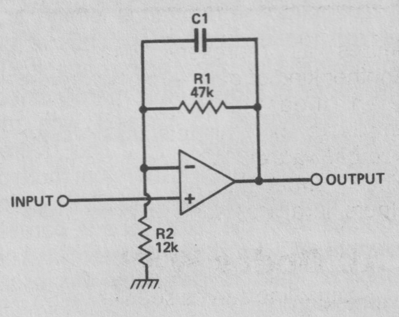


Figure 12...with a capacitor added in the feedback loop.

Figure 13 shows the required frequency response of the capacitor-resistor circuit formed by resistor R1 and capacitor C1, with a corner frequency at 20kHz. Now there is a magical formula which links the capacitance, resistance and frequency within a filter and this is:

$$f = \frac{1}{2\pi RC}$$

and so, by transposition, the value of

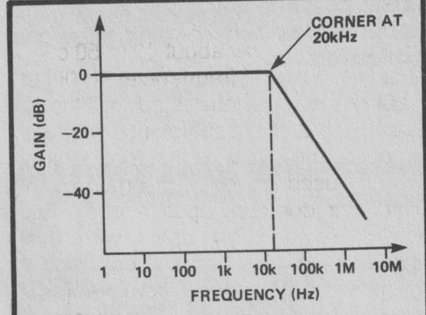


Figure 13. A graph of the frequency response required from the circuit in Figure 12.

of capacitor C1 is:

$$C = \frac{1}{2\pi fR}$$

and, inerting values:

$$C = \frac{1}{2\pi \times 20{,}000 \times 47{,}000}$$

$$= 1.69 \times 10^{-10} \text{ Farad}$$

$$= 17\text{nF}$$

And that's it! As simple as that. The frequency response for the amplifier with these values of components is shown in **Figure 14**.

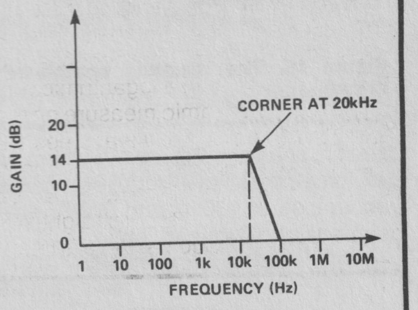


Figure 14. A graph of the response of the op amp circuit as modified by the 17nF capacitor.

Now, we can design the circuit so that it has a lower corner frequency at 100Hz. The idea is the same, we insert a capacitor-resistor circuit into the op-amp circuit so that the frequency is defined by the capacitor-resistor filter and the op-amp supplies the gain. The simplest way to do this for the lower corner frequency is to insert a capacitor between resistor R2 and ground as shown in **Figure 15**. The value of

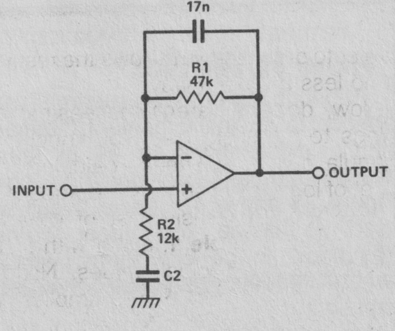


Figure 15. To add a lower corner frequency at 100Hz, a second capacitor is added.

capacitor C2 is defined in exactly the same way as before, using the same expression:

$$f = \frac{1}{2\pi\ RC}$$

so the capacitor value is given by the expression:

$$C2 = \frac{1}{2\pi\ fR} = \frac{1}{2\pi \times 100 \times 12{,}000}$$
$$= 0.000000133 \text{ Farad}$$
$$= 133 \text{ nF}$$

So the total circuit is shown in **Figure 16**, and the total frequency response for the circuit is shown in **Figure 17**, where we can see that the required corner frequencies of 100Hz and 20kHz have been attained. Not bad, eh?

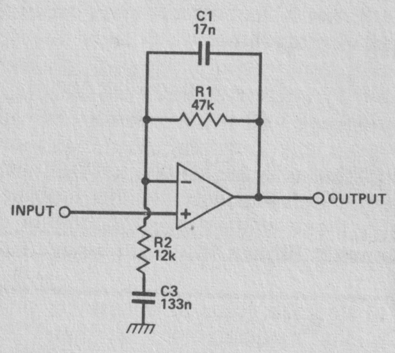


Figure 16. The resulting completed circuit...

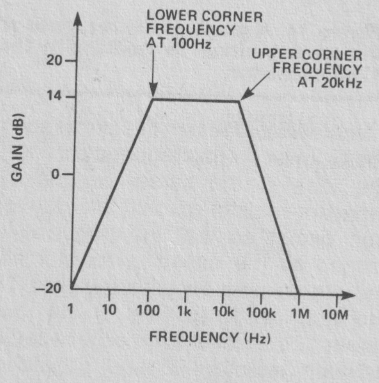


Figure 17...and the graph showing it frequency response range.

How It Works

Although this probably seems all very mystical to you at the moment, there are simple explanations to all of these effects, and we'll consider them now. **Figure 18** shows the basic circuit we used to define the upper corner frequency, in general terms, with a general frequency response curve. Let's explain it.

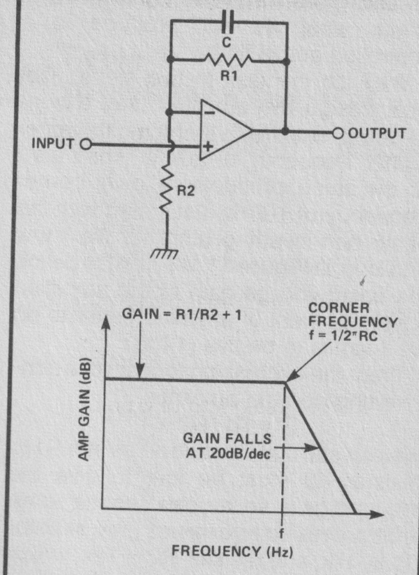


Figure 18. The basic frequency response defining circuit, with a graph showing the equations for calculating the gain and corner frequency, and the slope at which the gain falls off.

Any capacitor has a resistance. As we're looking at AC signals, though, we don't call it resistance, we call it **reactance** — just to make the difference clear. A capacitor's reactance is given by the expression:

$$X = \frac{1}{2\pi\ Cf}$$

You can see from the expression that it is dependent not only on the value of the capacitor, but also onthe frequency of the signal. To put it in plain terms, we can think of the reactance of a capacitor as being a resistance (which we can even use in, say, Ohm's law) the exact value of which depends on the frequency of the applied signal.

The capacitance reactance is inversely proportional to frequency, which means the reactance goes down as frequency goes up. So, at low frequencies, the reactance is high, and at high frequencies it is low. Thus, at low frequencies the effect of the capacitor in the circuit of **Figure 18** is merely that of high resistance in parallel with the feedback resistance R1. The overall feedback resistance is therefore more or less that of resistor R1. (Work it out for yourself from the resistance in parallel formula, if you don't believe it.) The amplifier gain is given by the usual expression:

$$G = Rf/R2 + 1$$

so the gain is what we would expect and this produces a level frequency response, as shown in the curve.

At high frequencies, on the other hand, the reactance of the capacitor is low. The effect is that of a low resistance (the capacitor) in parallel with the feedback resistor — so the overall feedback resistance is low. The gain of the amplifier thus falls, with increasing frequency, as shown.

The frequency where the changeover between level gain and falling gain occurs at (yes, you've guessed it) the corner frequency, which is defined by the expression:

$$f = \frac{f}{2\pi\ RC}$$

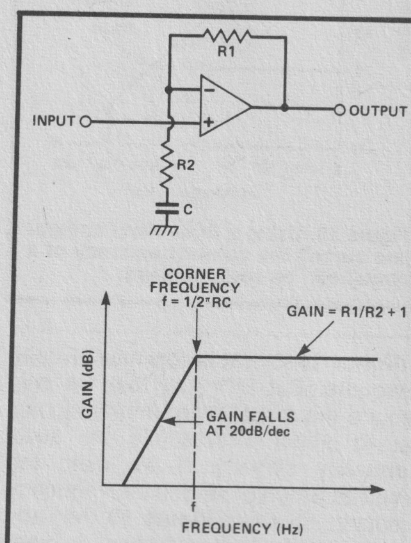


Figure 19. This circuit and graph performs the same function as those in Figure 18, but for the lower corner frequency.

In the case of the lower corner frequency definition, we can work through the circuit similarly. **Figure 19** shows the basic circuit and frequency response. This time the capacitor is in series with a resistor, so at low frequencies (when the capacitor reactance is high) the overall resistance (which equals resistor R2 plus the reactance of the capacitor) is also high. The gain is dependent on the value of this resistance and the higher it is, the lower is the gain. At low frequencies the gain is also low — the lower the frequency, the lower the gain. But at high frequencies, when the reactance of the capacitor is negligible compared with the value of the resistor, the overall resistance is effectively just that of the resistor R2. So at high frequencies the amplifier gain is level. As before the corner frequency occurs where:

$$f = \frac{1}{2\pi\ RC}$$

This is clearly shown in the frequency response curve of **Figure 20**.

More?

And we don't have to stop there. We can add more and more capacitor-resistor filters to the op-amp to do greater and greater things to the overall frequency response. **Figure 20** shows one obvious example, where instead of a parallel resistor and capacitor to define the upper corner frequency, a series of capacitor and resistor are used.

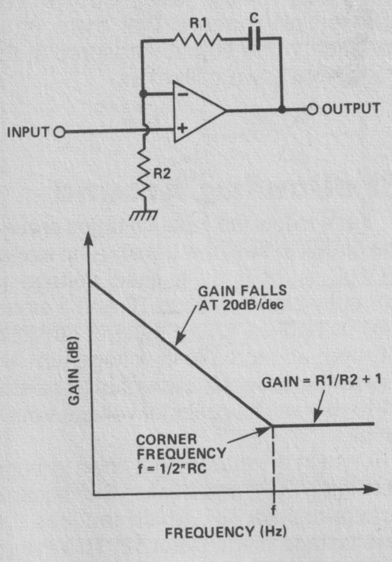


Figure 20. At high frequencies, the amplifier gain is level.

Figure 21 shows another, where the lower corner frequency is now defined by a parallel capacitor and resistor, instead of the series version we saw earlier.

You should be able to see that by choosing the values of capacitors and resistors, and also choosing their positions in the op-amp circuit, frequency responses of virtually any desired shape can be built up.

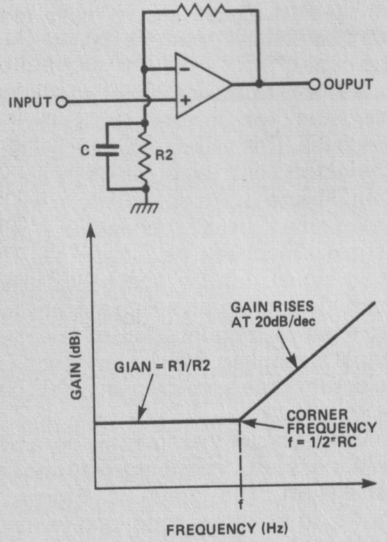


Figure 21. Here, the corner frequency is defined by a parallel capacitor and resistor, unlike the series pair in Figure 19.

Quiz

1. The reactance of a capacitor is equal to:

a the resistance of a resistor.
b the resistance of a transistor.
c resistance at one particular frequency.
d none of these.
e all of these.

2. The peak-to-peak voltage of a sinewave is:

a four times the amplitude.
b half the amplitude.
c equal to the amplitude multiplied by 0.5.
d b and c.
e all of these.
f none of these.

3. If a sinewave has a frequency of 2 Hertz:

a it has two rented cars.
b it has a wavelength of 0.5cm.
c it is not periodic.
d it is out-of-phase.
e it is periodic.

4. The amplitude of a periodic signal:

a has nothing to do with its wavelength.
b is another way of saying the size of the signal.
c is half the signal's peak-to-peak voltage.
d all of these.
e none of these.

5. A voltage gain of 10dB is the same as a voltage gain of:

a 0.316.
b 3.16.
c 31.6.
d –10dB.
e 3.16dB.

6. Frequency response curves can only show a maximum of two corner frequencies.

True or false.

Answers to last month's quiz:
1 a; **2** f; **3** d; **4** e; **5** True; **6** False.

Glossary Of Important Terms

amplitude the amplitude of an AC signal is the difference in voltage between its middle or common point and the peak voltage.

Bode plots method of showing a circuit's frequency response, where straight lines are used to approximate the actual response.

corner frequency the frequency in a frequency response plot at which the gain of the circuit changes from one value to another.

decibels logarithmic units of gain ratio.

frequency the number of cycles of a periodic signal in a given time. Usually frequency is measured in cycles per second, or Hertz (shortened to Hz).

frequency response curves graphs of a circuit's gain against the frequency of the applied AC signal.

Hertz the usual term for 'cycles per second'.

peak-to-peak voltage the difference in voltage between the opposite peaks of an AC signal. Commonly shortened to p-p voltage, or pk-pk voltage.

peak voltage the voltage measured at the peak of an AC signal.

periodic any AC signal which repeats itself regularly over a period of time is said to be periodic.

phase shifted two similar AC signals which are out-of-phase ie, start their respective cycles at different times are said to be phase shifted.

reactance a physical property of a capacitor which may be likened to AC resistance.

sinewave a constantly varying AC signal. Sinewaves are generally used to test electronic circuits as they comprise a periodic signal of one signal frequency.

wavelength the distance between two identical and consecutive points of a periodic waveform.

180° out-of-phase when two identical sinewaves are 'upside-down' with respect to each other.

COMPONENTS
FROM THE INSIDE OUT

Nicholas Bellenberg

Op Amps used to be discrete components and used in analogue computers. Thanks to integration, these circuits can be built into small packages and used as components in all kinds of applications.

OP-AMPS are analogue circuits (meaning that they're not digital, like nearly all modern computers are) and were originally designed to carry out mathematical operations in analogue computers. Analogue computers use voltages to represent data — digital computers use patterns of binary digits to do the job, but more of that another time. Although analogue computers are no longer used for pure computational purposes, the op-amp's computational ability is useful in control and monitoring systems that use actual voltages.

Op-amps were originally made from discrete (separate) components, but the development of integration has meant that these circuits can now be bought off the shelf, ready to slot into place and play useful roles in other designs.

Analogue computing circuits as mentioned, and other circuits like audio amplifiers, modulators, filters and so on can be constructed with op-amps that have different types of feedback circuit added to them. Op-amps are sometimes called linear circuits, as this word expresses their regulating action — as opposed to the switching action of digital circuits. As they are amplifiers, the function of op-amps is to increase the power, current or voltage that is applied to their inputs.

Two Inputs

A diagrammatical representation of an amplifier is shown in **Figure 1**. You can see that there are two inputs (+ and –). The corresponding op-amp symbol is shown in **Figure 2**. The op-amp has two inputs as well, a negative (inverting) input and a positive (non-inverting) input. This means that it is a **differential amplifier**. Its output is determined by the equation:

Vout = A (gain) × (Vin+ – Vin–)

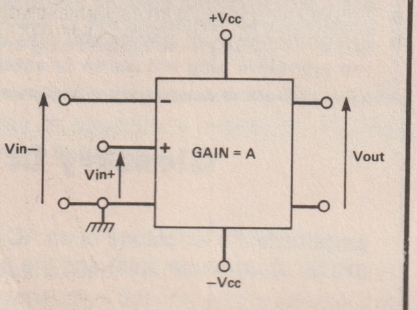


Figure 1. A diagrammatical representation of an amplifier.

It amplifies (makes larger) the difference between the two voltages at the two terminals. The gain of the amplifier is the **amount** by which it amplifies the input. All voltages are measured with respect to ground, so the input and output ground (0V) terminals are not shown on the diagram.

Inside the op-amp, the basic electronic circuit for the differential amplifier is similar to that shown in **Figure 3**. The two transistors are connected in what's called a **long-tailed pair**. It uses bipolar transistors in this example, but other types of transistors are often used in the actual ICs. One of the advantages of this type of circuit is that it remains very stable if either the temperature or supply voltage varies. Because both transistors are equally affected by such variations, then the way they are connected (differentially) means that there is no variation in the output measured at the transistors's two collectors.

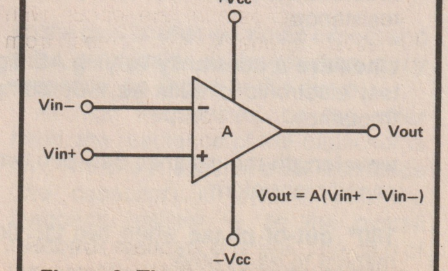


Figure 2. The conventional symbol for an op amp.

Following Around

Let's follow the input voltages around the circuit to see how the op-amp works. If Vin2 is held at a fixed voltage to correctly bias transistor T2 and a signal Vin1 is applied to the base of tansistor T1, then an increase in voltage Vin1 wil cause the collector current of transistor T1 to rise and its collector voltage Vout1, to fall.

A rise in the current that flows through transistor RT1 increases the voltage across resistor RE, which reduces the bias voltage of transistor T2. This in turn causes the collector current in transistor T2 to fall and the collector voltage to rise. The output voltage Vout2 is said to be 180° out of phase with the voltage Vout1. (See *Electronics From The Start*.)

This means that this output is the inversion or opposite of the other. In an actual op-amp both outputs are combined. In other words, a signal applied to Vin1 results in an output with an inverted sense — it is a negative, while a signal applied to input Vin2 gives an output of the same sense — so it is called non-inverted or positive. **Figure 4** sums up the various combinations of inputs and outputs that you can get with an op-amp. These should make clear the different inversions that take place in the circuit.

The key to the op-amps successful operation lies in the application of feedback. This allows very precise gain characteristics to be obtained for an op-amp. **Figure 5** shows the way in which an op-amp is usually connected — with negative feedback via resistor VF. The input signal can be applied to either input, depending on whether a positive or negative (non-inverted or inverted) output is required. Without any feedback an op-amp has a typical gain of 200,000 — an awful lot!

Another factor that limits the op-amps usefulness is its **frequency response** or **bandwidth**. The graph in **Figure 6** shows an op-amps open-loop (without feedback) gain as a function of frequency. It tails off dramatically as the frequncy increases. The gain is shown here in decibels (dB). The amplifier's bandwidth is determined as the frequency range over which the gain does not fall by more than 3dB.

We can see that the op-amp's open loop bandwidth in **Figure 6** is from 0 to 5Hz. Hz (standing for Hertz) is the conventional expression of frequency and corresponds to cycles per second. Normal healthy young ears can hear sounds in the frequency range 20Hz to 20kHz, deteriorating with age. So, we

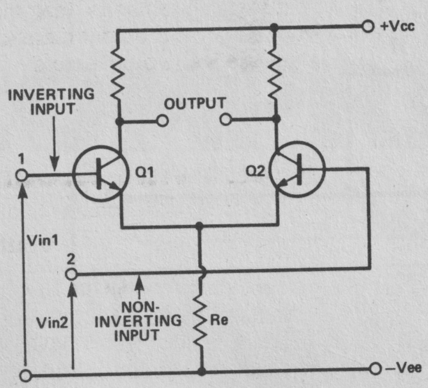

+Vcc
INVERTING INPUT
OUTPUT
1
Q1
Q2
Vin1
2
NON-INVERTING INPUT
Vin2
Re
–Vee

Figure 3. A circuit similar to that inside an op amp.

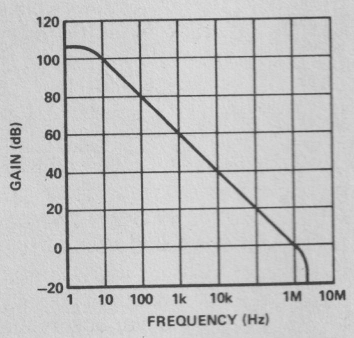


Figure 6. Gain according to frequency in an op amp open loop drops as frequency increases.

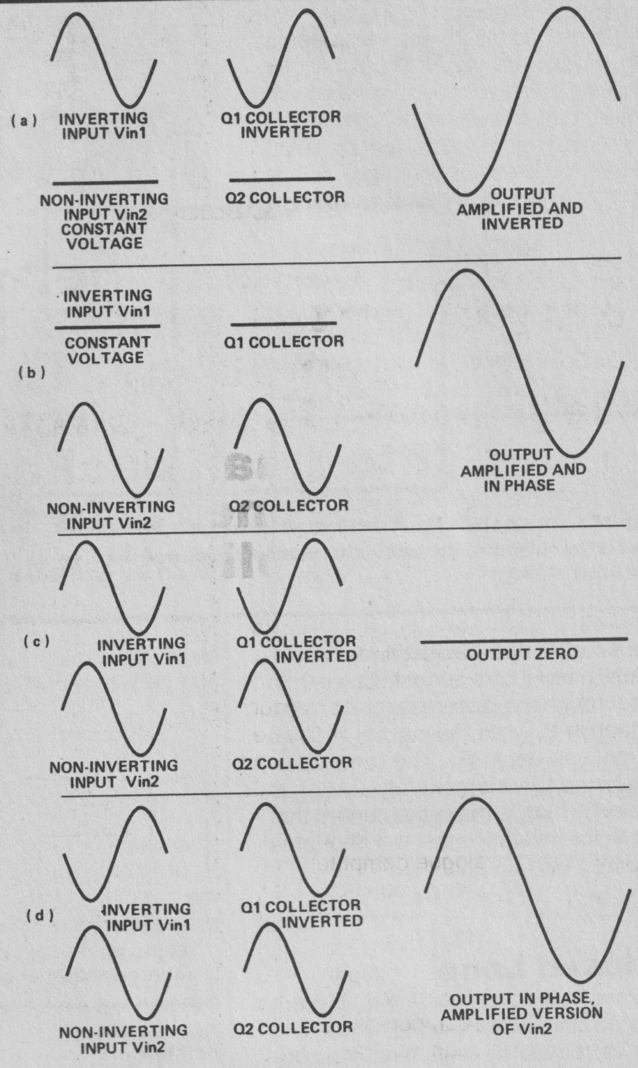

(a)
INVERTING INPUT Vin1
Q1 COLLECTOR INVERTED
NON-INVERTING INPUT Vin2 CONSTANT VOLTAGE
Q2 COLLECTOR
OUTPUT AMPLIFIED AND INVERTED
(b)
INVERTING INPUT Vin1
CONSTANT VOLTAGE
Q1 COLLECTOR
NON-INVERTING INPUT Vin2
Q2 COLLECTOR
OUTPUT AMPLIFIED AND IN PHASE
(c)
INVERTING INPUT Vin1
Q1 COLLECTOR INVERTED
OUTPUT ZERO
NON-INVERTING INPUT Vin2
Q2 COLLECTOR
(d)
INVERTING INPUT Vin1
Q1 COLLECTOR INVERTED
NON-INVERTING INPUT Vin2
Q2 COLLECTOR
OUTPUT IN PHASE, AMPLIFIED VERSION OF Vin2

Figure 4. Waveforms summarising the various inputs and outputs available.

can see that the op-amps open-loop frequency response would not be very good for an audio application. But we can improve the bandwidth — at the cost of sacrificing gain — by using feedback.

With the op-amp connected as in **Figure 5**, we have what is called closed-loop amplifier. The term feedback — as you've probably guessed — comes about because part of the output is fed back into the input. As this is negative feedback in this case the feedback goes to the negative input. The symbol G is used to represent closed loop gain. Aol represents open-loop gain. Feedback is of course proportional to the output voltage, so if we know what fraction of the output voltage the feedback is (called **β**) then we can work out the gain, G.

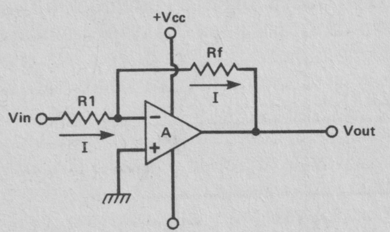

+Vcc
Rf
I
R1
Vin
I
A
–
+
Vout
–Vcc

Figure 5. An op amp connected as a closed loop amplifier.

Look at the op-amp circuit in **Figure 7**. We can see that it is connected for negative feedback. The input voltage:

$$V = Vin - \beta Vout$$

Vout =AolV, so:

$$Vout = Aol\,(Vin - \beta Vout)$$

right, pay attention at the back. If we now separate out Vout and Vin we get:

$$Vout\,(1 + \beta Aol) = Aol\,Vin$$

The effective gain of the circuit with negative feedback, G, comes from Vout/Vin:

$$G = \frac{Vout}{Vin}$$

$$= \frac{Aol}{1 + \beta Aol}$$

Easy. If there's no feedback, β will be 0 and the gain with feedback will be equal to the gain without feedback, Aol. The equation above tells us that the gain with feedback (G) is equal to the gain without feedback (A) divided by the term 1 -!β Aol, which is called the **return difference** or **feedback factor**.

Very Large

The expression for gain can be made simpler by remembering that Aol is very large, which means that the expression β Aol is very large too, depending on β. Then:

$$G = \frac{Aol}{\beta Aol}$$

$$= \frac{1}{\beta}$$

So if Aol is large enough, the gain depends on β. β is a fraction of the output voltage, and the fraction comes from the ratio of the feedback resistor value Re to that of the input resistor R1. The gain G:

$$G = \frac{-RF}{R1}$$

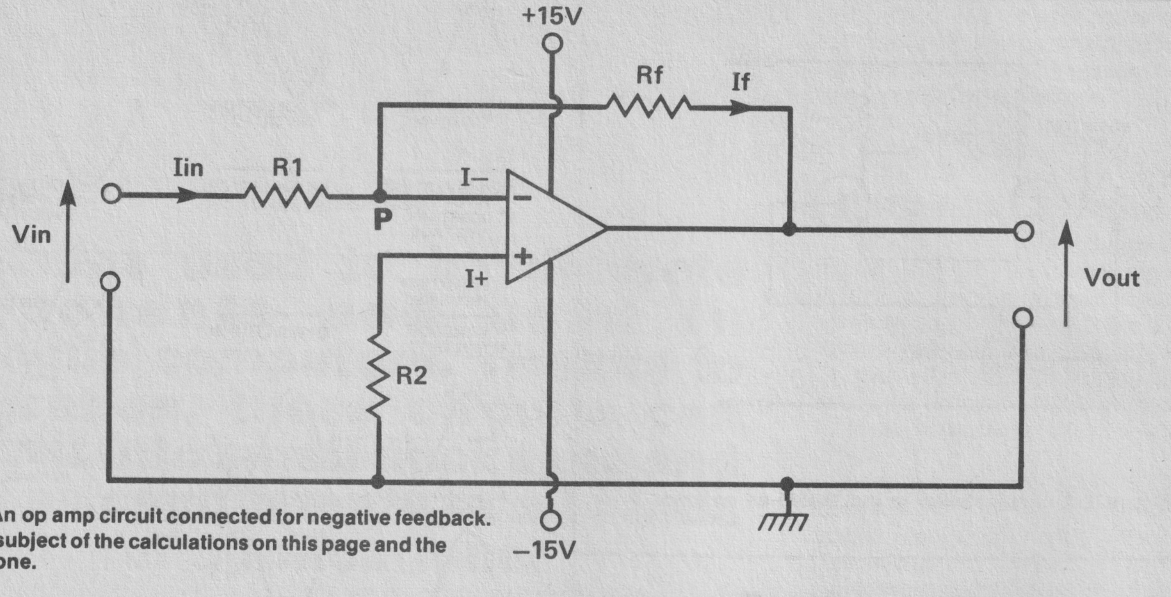


Figure 7. An op amp circuit connected for negative feedback. This is the subject of the calculations on this page and the previous one.

Let's work out the currents in the circuit to see if this is right. Current Iin flows through the input resistor R1 to the connection P, while the current IF flows through resistor RF. The amplifier's impedance (resistance to AC) is sufficiently high to make the curernt that flows in the inverting input (I–) negligible compared with Iin and IF.

Closed Loop

If we also assume the current going to the non-inverting input (I+) to be negligible as well, then the non-inverting input is at 0V. The voltage at the inverting input will be V, when measured with respect to ground. We therefore end up with:

$$\mathrm{Iin} = \frac{\mathrm{Vin} - \mathrm{V}}{\mathrm{R1}}$$

$$\mathrm{IF} = \frac{\mathrm{V} - \mathrm{Vout}}{\mathrm{RF}}$$

As Iin = IF, then:

$$\frac{\mathrm{Vin} - \mathrm{V}}{\mathrm{R1}} + \frac{\mathrm{V} - \mathrm{Vout}}{\mathrm{RF}}$$

Substituting –Vout/Aol for V (as V is the inverting input voltage, and the non-inverting input voltage =0):

$$\frac{\mathrm{Vin} + \mathrm{Vout}/\mathrm{Aol}}{\mathrm{R1}} = \frac{-\mathrm{Vout}/\mathrm{Aol} - \mathrm{Vout}}{\mathrm{RF}}$$

turning it all around will then give us:

$$\frac{\mathrm{Vin}}{\mathrm{R1}} = \frac{-\mathrm{Vout}}{\mathrm{RFAol}} - \frac{\mathrm{Vout}}{\mathrm{RFAol}} - \frac{\mathrm{Vout}}{\mathrm{RF}}$$

So what? Well:

$$\frac{\mathrm{RF}}{\mathrm{R1}}\,\mathrm{Vin} = -\mathrm{Vout}\ \frac{\mathrm{RF}}{\mathrm{R1Aol}} + \frac{1}{\mathrm{Aol}} + 1$$

Now the closed-loop gain, G, can be seen to be:

$$\mathrm{G} = \frac{\mathrm{Vout}}{\mathrm{Vin}}$$

$$= \frac{-\mathrm{RF}/\mathrm{R1}}{1/\mathrm{Aol}\,(\mathrm{RF}/\mathrm{R2} + 1) + 1}$$

If Aol is big enough, then we can ignore the term multiplied by 1/Aol, as this will be a very small number. We therefore end up with a closed loop gain of:

$$\mathrm{G} = \frac{\mathrm{Vout}}{\mathrm{Vin}}$$

$$= \frac{-\mathrm{RF}}{\mathrm{R1}}$$

This value is negative, as the inverting input is being used. It also proves that the gain, G, is determined by the ratio of the value of the resistors RF and R1.

As the gain Aol is very large (because V = Vout/Vol), V must be tiny in comparison with Vout. So the inverting input can be thought of as being at 0V, if Aol is large enough. The connection point P in **Figure 7** — whilst not actually being directly connected to ground — is virtually at 0V, and so is called a **virtual earth**. And not a lot of people know that!

So, let's bung some values into this op-amp's components and see how all this theory comes out. Suppose we want the circuit to give a gain of –10 (negative, remember, because the output is inverted), and that R1 is 100kR:

$$\mathrm{G} = \frac{\mathrm{RF}}{\mathrm{R1}}$$

RF therefore:

$$= 10 \times 100\mathrm{k}$$

$$= 1\mathrm{M}$$

$$\mathrm{R2} = \frac{\mathrm{RF\ R1}}{\mathrm{RF} + \mathrm{R1}}$$

$$= \frac{1 \times 10^6 \times 1 \times 10^5}{1 \times 10^6 + 1 \times 10^5}$$

$$= \frac{1 \times 10^{11}}{11 \times 10^5}$$

= (approx) 100kR

which is the nearest actual value of resistor available. So, if 100mV is applied to the input, the output will equal:

$$10 \times -100\mathrm{mV} = -1000\mathrm{mV}$$

$$= -1\mathrm{V}$$

Earlier we spoke of bandwidth, and the curve at the top of **Figure 8** shows an unaltered open-loop op-amp's bandwidth. You can see that the curve for the closed-loop op-amp has a much lower gain and a much improved bandwidth.

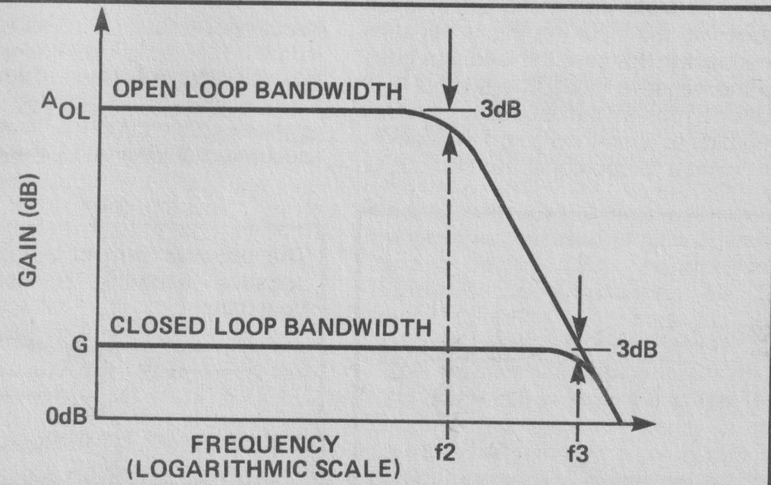
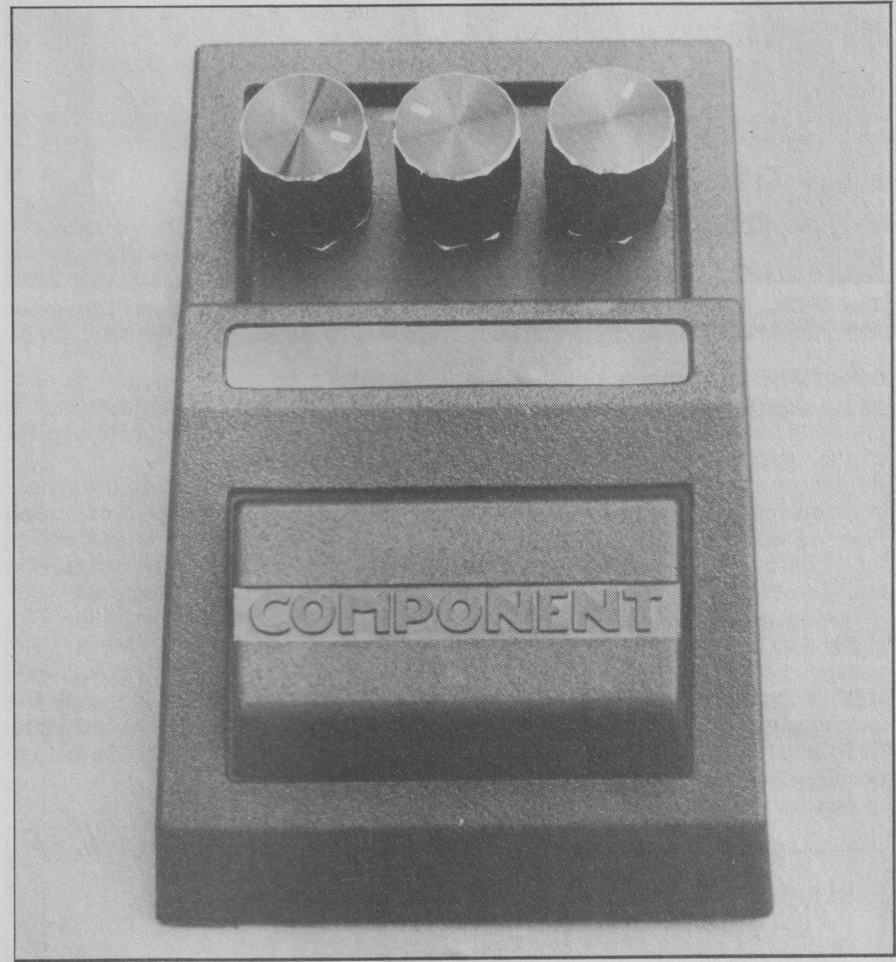


Figure 8. Bandwidth curves for open loop and closed loop op amps: the closed loop has a lower gain and an improved bandwidth.

EM MUSIC EFFECT COMPRESSOR PEDAL



More note-per-pick with an effect which allows your riffs to run and run.

A Armstrong

ONE OF THE more commonly used guitar effects is a distortion free sustain. The gadget which provides this effect is known as a compressor.

The effect of such an effects pedal may be imagined as that of a man using a volume control to keep the sound level more or less constant as the string vibration dies away. Of course, the man would have to be very fast on the control to produce the right effect, but luckily electronic circuitry does not suffer from the same limitation on speed or reflexes.

This project is a compressor with several options which may be selected at the time of construction. It is built into a footswitch box intended for use with guitar effects. The case chosen has a substantial foot pedal at the front, connections at the rear, and a recessed control panel to give the controls some protection from the heavy footed.

Attack!

The speed at which this compressor turns down the volume to compensate for a loud note, the attack, is adjustable in this design, as is the decay time. The decay time is, of course, the rate at which the compressor turns up the volume again if the strings are suddenly damped.

The most essential control though is the compression control, which controls the amount by which a fading note from the guitar is boosted. Clearly, the compressor cannot simply increase the amplification to keep its output exactly the same as the input signal dies away. Eventually, the noise and hum from the guitar pickup would be the predominant signal. Therefore, a control is provided to adjust the degree of compression used.

How It Works

Figure 1 shows the basic principle of a compressor. The rectifier and voltage controlled amplifier form a negative feedback loop, which tends to keep the output signal level constant. The loop gain, however, is fairly low, so that amplitude variations on the input signal are simply reduced on the output. The more detailed block diagram of this project is shown in **Figure 2**.

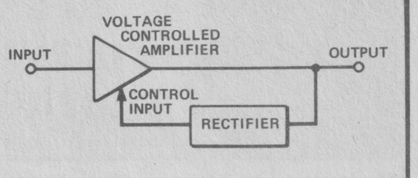

VOLTAGE CONTROLLED AMPLIFIER
INPUT
OUTPUT
CONTROL INPUT
RECTIFIER

Figure 1. The principle: a feedback loop which keeps the output constant.

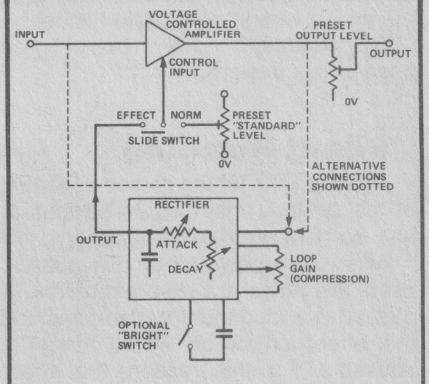
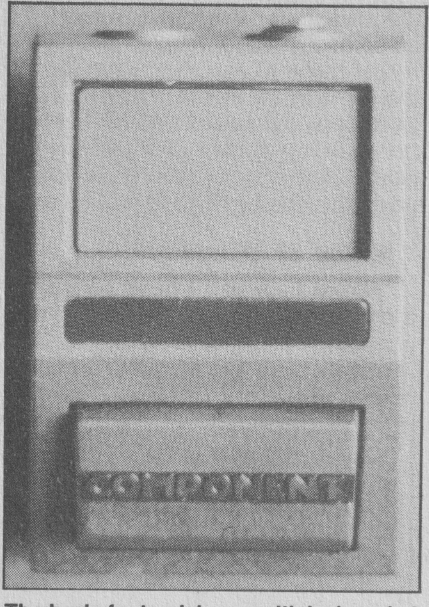

VOLTAGE CONTROLLED AMPLIFIER
INPUT
PRESET OUTPUT LEVEL
OUTPUT
CONTROL INPUT
0V
EFFECT
NORM
PRESET "STANDARD" LEVEL
SLIDE SWITCH
0V
ALTERNATIVE CONNECTIONS SHOWN DOTTED
RECTIFIER
OUTPUT
ATTACK
DECAY
LOOP GAIN (COMPRESSION)
OPTIONAL "BRIGHT" SWITCH

Figure 2. A block diagram. Some of the blocks are optional.

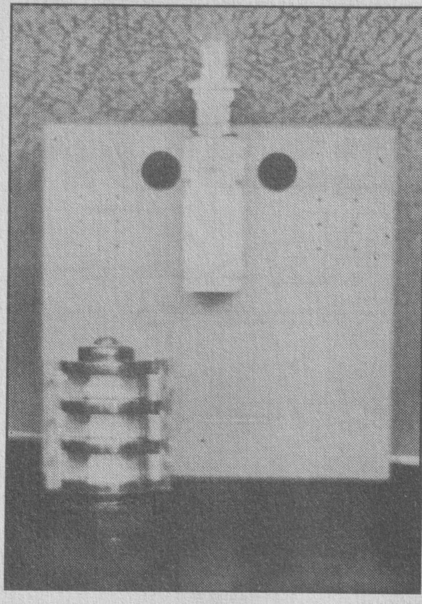
This shows the same basic parts as **Figure 1**, and details of other essential parts of the circuitry required for the functions of this project. There are some choices to be made at the time of construction, and once the function of the different circuit blocks is clear, the basis for choosing the options will also be clear.

The first point to note here is that there are two possible connection points for the rectifier. As well as the conventional connection to the input, to adjust the level by feedforward control. Using this connection, the output level may be arranged to decrease as the input level increases, and vice-versa.

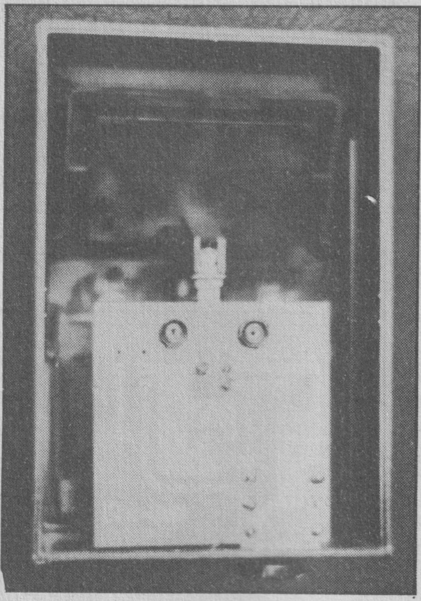
Another option is to add a 'bright' switch. What this does is to restrict the performance of the rectifier at high frequencies, so that loud treble notes



The basic footpedal case with jack socket set into the rear.



The PCB, spring switch and jack connection supplied with the Maplin case.



The track side of the original PCB; we are supplying our own PCB design.

come through at almost full volume, rather than having their volume reduced. If the rectifier has low gain at high frequencies, then when only a high frequency note is present there will be little input to turn down the gain of the voltage controlled amplifier. This is to compensate for an apparent dulling of the sound when compression is used.

This sums up the major circuit areas used in this design.

Silent Running

The reasons for the choice of certain of the controls may not be obvious at first, so here are the main reasons.

When high powered amplification is used, any noise is a severe problem. In particular, a loud thump when switching the effect on or off would render it unusable. For this reason, the effect on/off switching is carried out on the control line rather than on the signal itself. When the effect is off, the voltage controlled amplifier is switched to a preset gain. There is just sufficient time constant to avoid audible effects, other than a small change in volume, which is inevitable.

In order to avoid a dramatic change of gain when switching from effect to normal, it is of course necessary to run the amplifier at a moderate gain in the normal position. This in turn means that an output level control is required so that the gain in the normal mode can be set to unity. In this project a preset control is used, because there is a limited space available for controls.

The effect/normal switch itself is a latching type, so electronic latching is not required. This sums up the main controls required for the unit.

The Circuit

The building blocks from which the circuit is made are of a conventional nature. The interesting aspects of the circuit design are the reasons for the choice of building block, and also some of the sniggles and contrivances, or design points which affect the performance in not-so-obvious ways. First, we will take a general look at the functional blocks. The circuit diagram is shown in **Figure 3**.

The obvious place to begin is with the signal path from input to output, via the voltage controlled amplifier. The signal is AC coupled to the input of op-amp IC1, which is biased to half the supply voltage by R1 and R2. There is a high frequency decoupling capacitor, C10, at this point to prevent the circuit acting as a radio receiver. The amplifier circuit is non-inverting, and provides a gain which is determined by the ratio of R3 and R4 plus the resistance of Q1.

The FET, Q1, is of course the control element, whose resistance is controlled by the voltage supplied to its gate via R6. The lowest gain this circuit can provide is unity, if the FET resistance is infinite. The highest gain possible, if the FET resistance is zero, is (R3+R4)/R4, which, with the component values in use, is 22.36363636, assuming that the resistor values are perfectly accurate. What this means in practice is that the

Parts list

RESISTORS

(All ¼W or great er carbon)

R1,2	100k
R3,8,11	47k
R4,9	2k2
R5,6	1M
R7,10	10k

POTENTIOMETERS

RV1,2,6	47k preset (smaller than standard size: eg, Maplin's cermet.)
RV3	100k carbon track linear
RV4	47k carbon track linear
RV5	10k carbon track linear

CAPACITORS

C1,2	2u2 radial electro
C3,4,5	47u radial electro
C6,8	100n polyester 0.2 or 0.3 inch pin spacing
C7	10u radial electro
C9,10	1n ceramic (or anything 0.2 inch pin spacing)
C11	4n7 (optional)

SEMICONDUCTORS

D1	1N4148
Q1	2N3819 or almost any ordinary N channel junction FET
IC1	5534 low noise op-amp
IC2	LM358 dual op-amp (*only* alternative: CA3240)

MISCELLANEOUS

SW1 single PCB press switch (supplied with footswitch box); optional SW2 miniature on off switch; PCB mounting stereo jack (supplied with footswitch box); PCB mounting mono jack socket; 3 knobs; single footswitch box from Maplin; battery clip to fit PP3; PP3 battery; wire, solder etc.

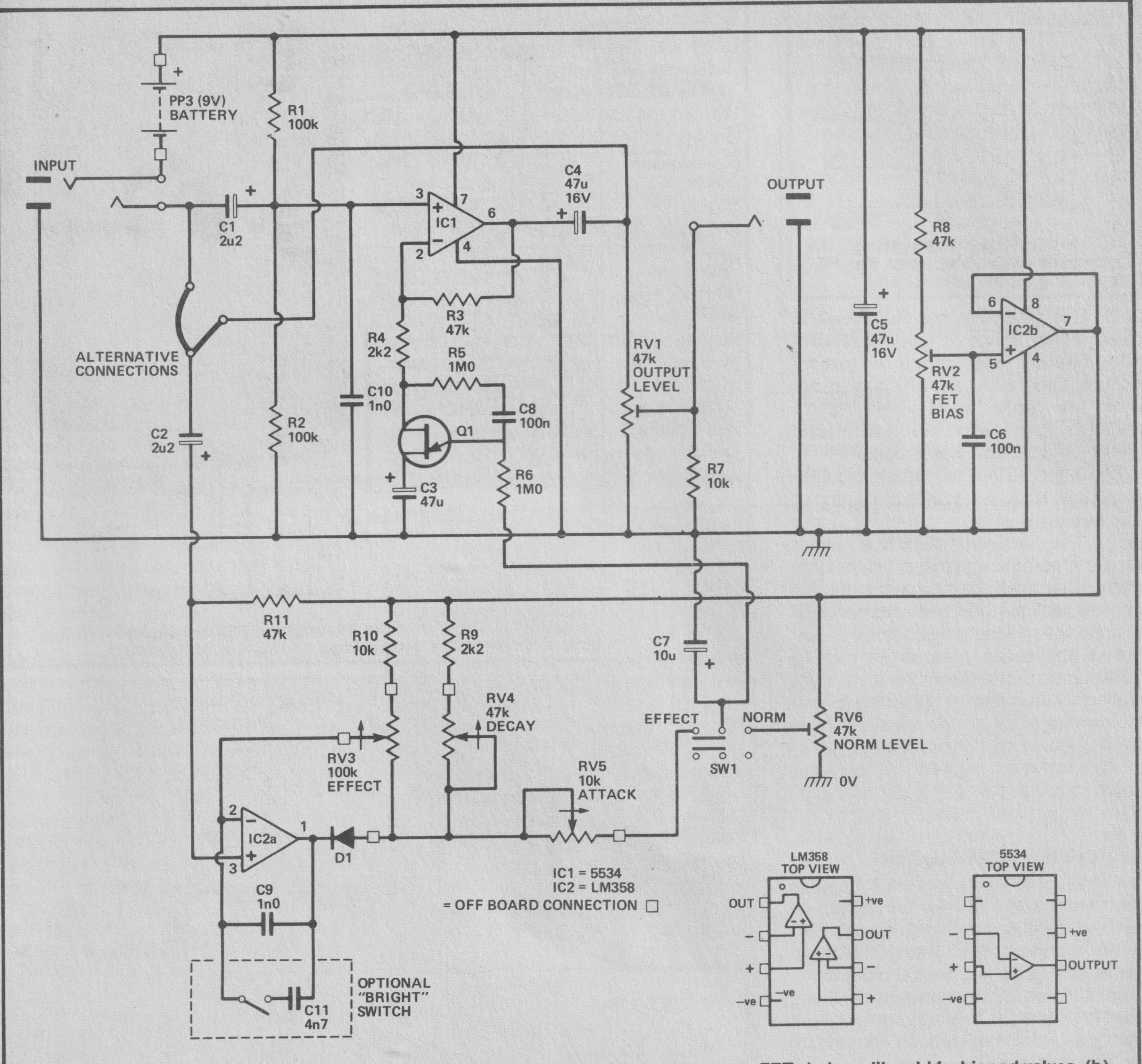

PP3 (9V)
BATTERY
R1
100k
INPUT
C1
2u2
ALTERNATIVE
CONNECTIONS
C2
2u2
R2
100k
IC1
C4
47u
16V
R3
47k
R4
2k2
R5
1M0
C10
1n0
C8
100n
Q1
C3
47u
R6
1M0
RV1
47k
OUTPUT
LEVEL
R7
10k
OUTPUT
C5
47u
16V
R8
47k
RV2
47k
FET
BIAS
IC2b
C6
100n
R11
47k
R10
10k
R9
2k2
C7
10u
RV4
47k
DECAY
RV3
100k
EFFECT
RV5
10k
ATTACK
EFFECT
NORM
SW1
RV6
47k
NORM LEVEL
0V
IC2a
D1
C9
1n0
C11
4n7
OPTIONAL
"BRIGHT"
SWITCH
IC1 = 5534
IC2 = LM358
= OFF BOARD CONNECTION ☐
LM358
TOP VIEW
OUT
–
+
–ve
+ve
OUT
–
+
5534
TOP VIEW
+ve
OUTPUT

Figure 3 (a) The circuit: the individual design blocks are straightforward. In some ways FETs behave like old fashioned valves. (b) The internal circuit of the LM358 and 5534 ICs.

gain of this stage can be controlled over a range of about 20:1.

The output signal from the voltage controlled amplifier is AC coupled to the output level control, the need for which has already been explained. The output load resistor, R7, modifies the law of the control to give a rough approximation to a logarithmic characteristic. Log presets are not available, so this is the best that can be done. In any case, a perfect volume control law is not required, but a non-touchy preset adjustment is helpful.

The rectifier circuit, which uses IC2b, can take its input from the output or input signals of the unit, as mentioned before. The effect of this is discussed in the boxed section. The circuit used is a half wave precision rectifier, which is designed to give a negative going output relative to a presettable bias voltage. The gain of the circuit is adjusted by RV3, and the charge and discharge times of C7 can be adjusted by RV5 and RV4 respectively. Again, the effect on the sound is discussed in the boxed section.

The rectifier's zero point voltage is provided by IC2b, and adjusted by RV2. This is a method of providing a voltage which does not vary as C7 charges and discharges, without passing a large steady current through the setting pot. In a battery powered circuit this is an advantage, though in a mains powered circuit, the voltage could be provided by a 1k pot without the intervention of IC2b. This sums up the circuit overview and leads to the more detailed considerations.

Design Details

There are two aspects which need a more careful consideration, the control of gain via the FET resistance, and the slew rate of the precision rectifier. The FET is the logical place to start.

The FET used is of a type which is symmetrical. That is to say that the souce and drain terminals are interchangeable. Now it is well known that a FET will work as a resistor if only a small voltage is applied between source and drain. The resistance depends on the gate voltage, as shown in **Figure 4**. The fact that it is symmetrical means that it can be made to work in either direction, hence it will work with an AC signal.

To make it work genuinely symmetrically, it is necessary that the gate voltage is controlled relative to the middle of the channel in the FET, rather than relative to the source of the drain. R5 and C8, working as a potential divider with R6, ensures that, from the point of view of AC, the gate voltage is halfway between the source and the drain, even though its DC potential is negative of

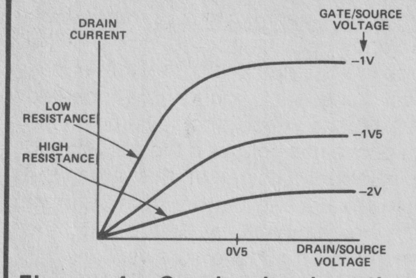


Figure 4. Graph showing the 'resistance' available from the FET depending on the gate voltage.

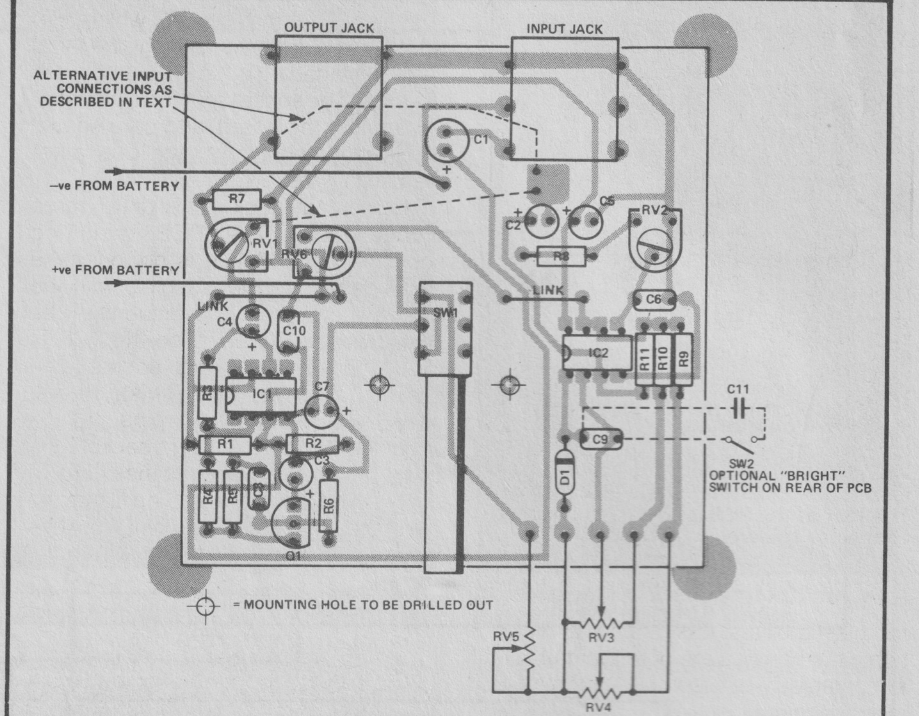
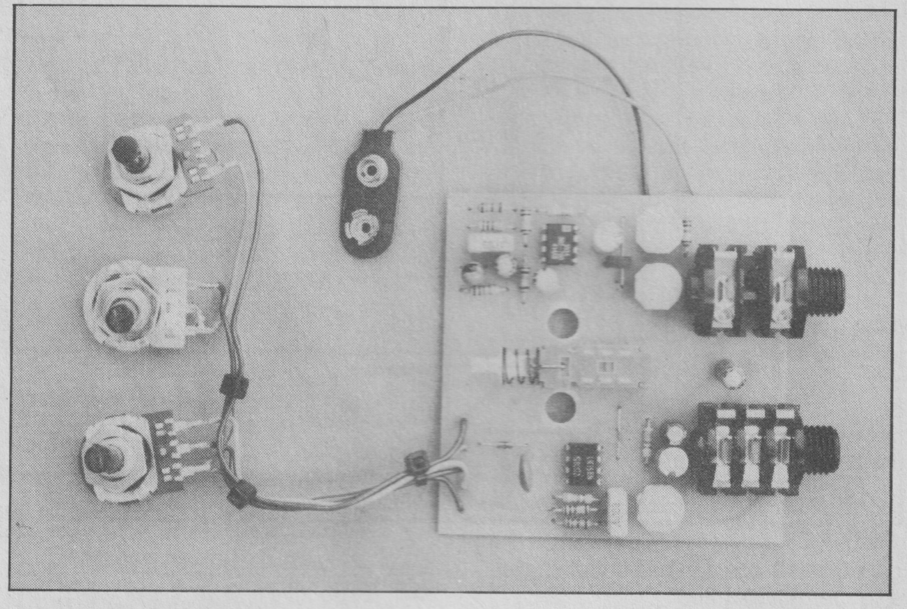


Figure 5. The layout of the PCB, showing the original SW1 as supplied with the case, plus a number of alternative connections on the rear of the board.



The PCB made up with hole drilled for the pillars, and the pots connected.

both. This gives the maximum peak to peak signal range before significant non-linearity, and hence distortion, occurs. Otherwise, the FET would not work the same for positive and for negative signal excursions, and second harmonic distortion would occur.

Some people may wish to leave out R5 and C8 in order to simulate the sound of an overdriven valve amplifier input stage. The functioning of FETs is indeed very comparable with that of valves. The main differences are the voltage ratings, and the fact that FETs do not use little electric fires inside glass bottles.

The application of a larger voltage from source to drain results in a constant current characteristic, but in this circuit the applied voltage is only equal to the signal input voltage, because the signal on the inverting input of the op-amp tracks that on the non-inverting input, even though the output signal may be rather higher in amplitude. Guitars with very high output pickups may drive the circuit to produce some distortion, but it is unlikely to be severe.

The rectifier zero voltage provides the standing bias on the FET. It is adjusted to the point at which the FET resistance just starts to increase above its minimum possible level. In this way, any output from the rectifier will reduce the gain of the amplifier stage.

Slew Rate

The rectifier circuit is purely half wave, and, in order to avoid any extraneous effects on the 'Decay' time constant, there is no DC feedback to the negative input when D1 is reverse biased. This means that C7 cannot discharge through the wiper of RV3, but it also means that the op-amp output crashes into the power supply rail with a resounding click. So abrupt is this effect that it can be picked up on the sound, and give an unwanted fuzz effect. Capacitor C9 gives AC feedback, and reduces the slew rate of the output under these circumstances to something less harmful to the sound.

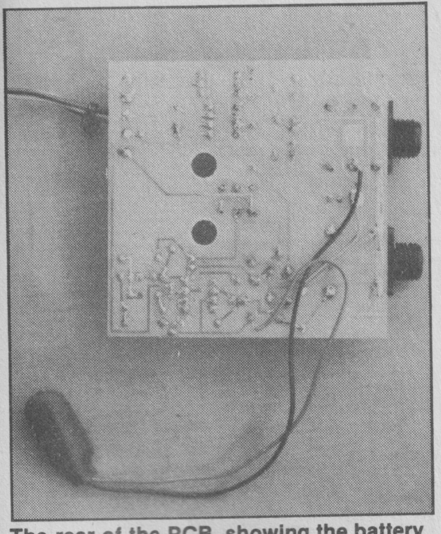
During the testing of the prototype, the initial choice of value for C9 prevented the rectifier circuit from responding significantly to treble notes. A suitable value was chosen, but this initial problem gave rise to the idea of a 'bright' switch, which would prevent treble notes from being compressed. This prevents the 'squashed' sound that may otherwise occur. Thus, SW2 and C11 were added to the circuit diagram, but C11 must be off board, and this is purely an optional addition for those who want it.

If the circuit is driven into clipping, RF oscillation may start. This is self maintaining once started. The remedy is to connect a 47pF capacitor between pins 5 and 8 of the 5534. This may be done on the back of the PCB or on the pins of the IC above the board.

Well that about wraps it up for the design. All important facets have been mentioned. Now it's time to build the project.

Construction

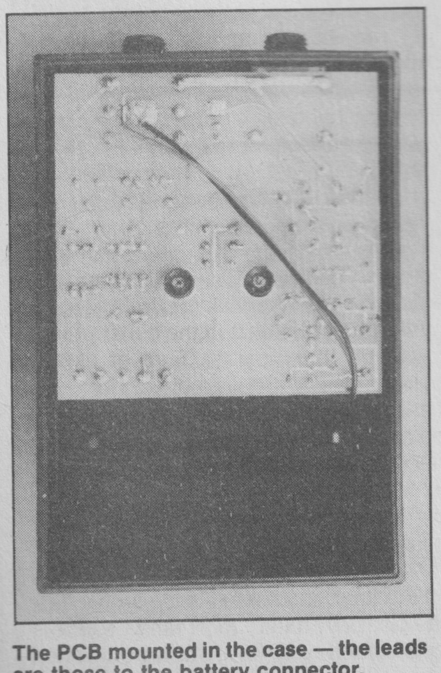
Because this project is designed to fit a specific foot pedal case, care is needed over certain aspects of the construction to ensure a proper fit. First of all the bottom should be removed from the case, and the PCB inside it should then be removed. To do this it is also necessary to unscrew the plastic nut which retains the jack socket on the rear of the case. The PCB may then carefully be lifted off the plastic pillars on which it is supported. If it is a particularly tight fit a little plastic may need to be shaved off the pillars by means of small wire cuters or a modelling knife. If this is necessary, remove the minimum amount of plastic.



The rear of the PCB, showing the battery connector soldered to the board.

The mounting holes on the project PCB should now be drilled out to fit over the plastic pillars. It is best to aim for a tight fit. When you are sure that the fit is acceptable, the assembly of the PCB may proceed.

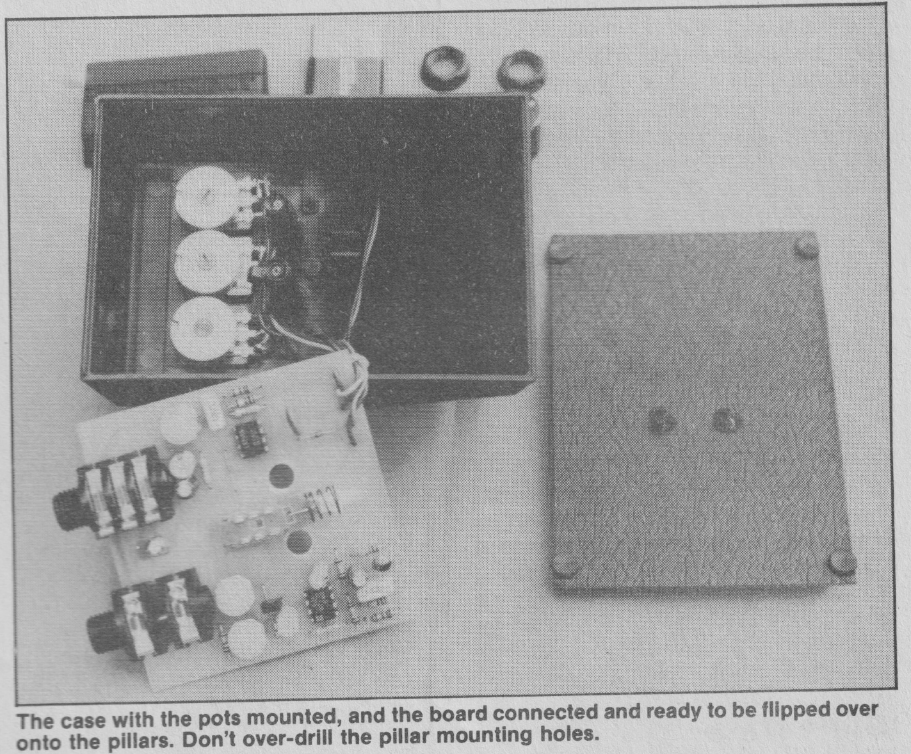
As is normal practice, the capacitors and resistors and wire links should be fitted first, followed by the semiconductors. Before fitting the FET, check on its pin connections. Some devices have the gate terminal in the middle, and some have it at one end. The PCB is laid out with the gate connection in the middle. If the particular FET which you are using has its gate terminal at one end, then this terminal should be connected to the middle pad and the other two terminals connected to the other two pads — it does not matter which way round. So long as the gate terminals of the FET is connected to the middle pad, it does not matter which way round the other two are connected, as may be clear from the description of the functioning of the circuit.



The PCB mounted in the case — the leads are those to the battery connector.

Now it is time to fit the switch and the sockets. The input socket, which is a stereo jack, and the switch are both items salvaged from the PCB supplied in the case. The socket should be mounted flat against the board, and care should be taken to mount the switch level rather than slightly askew.

The output jack is slightly more problematical. The PCB mounting socket sold by Maplin is not quite the same design as that supplied with the foot switch box. Consequently, the positioning of the holes on the PCB is slightly different for this socket. The difference does not finish there, however, The mounting pins have a shoulder which prevents the socket from being mounted flat against the board. In order to make the whole thing fit into the case, the shoulders of each pin must be trimmed so that the socket will fit flat against the PCB. A small pair of wire cutters should accomplish this with little difficulty.



The case with the pots mounted, and the board connected and ready to be flipped over onto the pillars. Don't over-drill the pillar mounting holes.

The three control pots should now be connected up as shown in the diagram, and the battery connector should be connected temporarily to the appropriate pads on the PCB. The remaining thing to connect is the wire link which supplies signal to the input of the rectifier circuit. This is mounted on the back of the PCB in one of the two positions shown, according to whether you want to use feedback or feedforward compression. If later experimentation shows that this connection needs to be changed, the fact that it is on the back of the board means that it can be moved without removing the board from the box.

A LED which shows when the effect is engaged may be helpful. The spare half of the switch may be used to turn on a LED — in order to save current use a 4k7 resistor in series with the LED, and a high efficiency LED. Under the middle of the black shiny trim along the top of the case, there is no round hole moulded in the plastic. A hole should be cut in the trim above this hole, and the LED should be inserted. To get at this area, the control plate and battery compartment must be removed, by unscrewing the two internal mounting screws.

The 0V supply for the LED should of course be drawn from the circuit side rather than the battery side of the input socket, so that it will be disconnected when the rest of the circuit is switched off.

Testing

Before mounting the board in the box, it should be tested. The power is switched by one of the contacts of the input socket, so a plus should be inserted, and a battery connected. The first job is to measure the voltage on the output pin of IC1. Measure between pin 4 and pin 6, and check that there is approximately 4.5 volts present. If the voltage is significantly below 4V5 then first check the battery voltage. If it is low, replace it and try again.

If the output voltage is not correct, look for a short circuit, and check that C1, C3 and C4 are inserted the correct way round. Most possible faults in the voltage controlled amplifier section should now have been found.

The next part to examine is the FET bias line. Using the voltmeter, make sure that the voltage on pin 7 of IC2 can be varied between 0V and 4V5 by adjusting RV2, then leave this set to 4V.

Now, plug the other end of the input lead to the guitar if you haven't done so already, and press the switch in. Twang a string while measuring the voltage on the switch end of R5. The voltage here should start at about 4V, and should fall when the string is plucked. If it does not, the most likely cause is misconnection of the potentiometers, so check this first.

When the rectifier responds to signals, everything should be working, and it is time to try out with an amplifier. Connect the effect to the power amplifier, making sure that the amplifier volume is set to a low level. The next task is to set the FET bias.

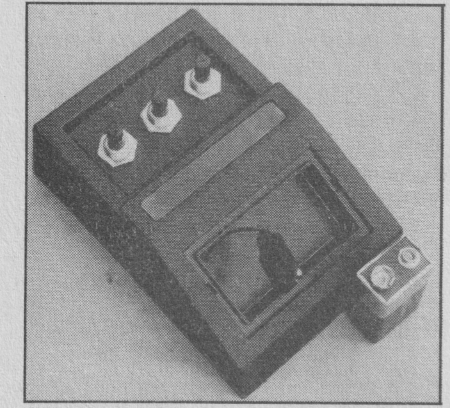
Release the switch, so that the effect is off, and set RV6 so that the wiper is at the opposite end from 0V. Set the output level control, RV1 to half way. Now, while repetitively plucking a string, set the FET bias until there is a very slight reduction in sound level, and then back it off so that the sound level is just not reduced. Twang a bass string fairly hard, and listen to how it dies away. Switch the effect on, and set the attack pot fully clockwise, the decay pot anticlockwise, and the compression pot clockwise. Try the test again, and hear the difference in how fast the note dies away.

Now tweak the output level control to the level desired. Note that if the compressor is to drive a significantly capacitive load, such as a long lead, then it may oscillate at radio frequency if the output level is set to its maximum.

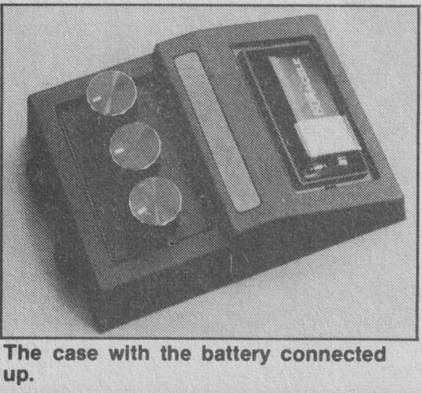
Boxing

Now that the effect is working, it needs to be fitted into its box. All that has to be done to achieve this is to drill holes for the pots and for the output socket. The positioning of the hole for the output socket is critical, and the PCB is laid out so that if it is drilled at the same height as the other hole, and in the middle of the area set aside for it, the socket will fit into the hole. In practice, it will probably be necessary to drill the hole oversize, and rely on the plastic bezel to hide the fact. Bodgers of the world unite.

Before actually fitting the board, remove the battery connector and reconnect it with the wire threaded through from the battery compartment. In case you haven't noticed it yet, this is under the pedal, and to get at it you simply lift off the plastic cover. Now the final assembly can be completed.



The case with the battery clip in place.



The case with the battery connected up.

Applying The Effect

COMPRESSION can be a useful effect, but there are several points which must be borne in mind in order to use it well. First of all, we will consider what it does to the signal, and its limitations.

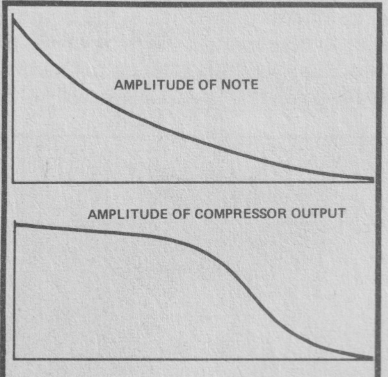


Figure 6. A graph of a normal guitar note's decay, compared with that of a compressed note.

Figure 5 shows the effect of a compressor on a note dying away. As can be seen, once the note has decreased to a very low level the gain of the compressor can increase no more, and the output dies away. This points up a fairly obvious limitation to this effect. If the guitar has no sustain to speak of, then the compressor cannot magically create it. It can only enhance sustain which already exists to some extent.

Another obvious limitation of a simple compressor is that it cannot distinguish between noise and a very low level signal which must be amplifed with maximum gain. As a result, the noise generated by any effect units connected to the input of the compressor will end up very loud. A compressor should normally be the first effect in the chain if several are patched together. On the plus side, a compressor used before other, noisy, effects, can reduce their effective noise level. This is because the signal fed into these effects is maintained at a higher level for most of the time.

The speed of response of a compressor can have a profound influence on the overall effect. If, for example, a loud chord is struck, and the attack is not fast enough, it is possible for the output to overload or clip before the gain is reduced sufficiently. An example of a waveform exhibiting this problem is shown in **Figure 6.** This gives the sound of a pop as a loud note is struck.

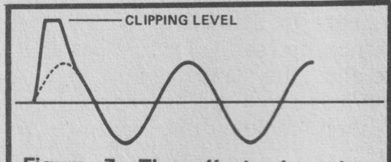


Figure 7. The effect of a slow response from a compressor circuit: the initial note may clip.

The converse effect is that if the attack is set very fast, and the decay rather slow, the actual striking of a note can reduce the gain too far. It will then recover gradually during the note, and for the first portion of the note it will actually get louder. This can be quite an interesting sound, and can be accentuated by using the input signal connection to the rectifier.

Up to now, all comments have been related to guitar applications, but a compressor can be useful with a microphone as well. A moderate degree of compression can accentuate the vocals, or partially compensate for a poor microphone technique. Too much compression will almost guarantee feeback, so only use the compression knob sparingly. A fast attack and a medium setting of decay is likely to be most suitable for this application.

In general, to avoid the drawbacks of compression, only use as much as is required, and do not over-compress.

Shop

Most of the components used are easily obtainable, but pay particular attention to the author's comments about component sizes. RV1, 2 and 6 need to be a slightly smaller size than the commonest carbon track pots.

The footpedal case used comes from Maplin (0702 554155) and costs £5.95, although there is nothing to prevent you from adapting the board size and switches for other case types.

The components should cost around £7 excluding case and PCB. The PCB is not available from Electronics Monthly at present, but will be when the PCB Service resumes.

AXING INSIDE QUESTIONS

Marcus Smith

What can you do with an electric guitar— apart from play it? We ask— I mean ax— the pros some questions.

A CLOSE COLLEAGUE of mine has a guitar. Every now and then it appears in the corner of the office, like a papoose. Normally it stays there.

One brief encounter with this instrument told me why she wasn't keen to show it off. Putting it simply, the neck doesn't fit, the bridge is at the wrong angle and there is virtually no output from the top E string.

'I keep asking people ' she said, 'They all say "buy a better guitar".

I assume that the EM reader wants to know, firstly, what can be done for an instrument in the safety of his own workshop, and secondly, how to find out more about the subject.

With this aim I went to Andy's Guitar Workshop, makers and repairers of stars' guitars as well as ordinary people's guitars, and talked to Malcolm Miles, who, with his wife Fiona, has been repairing guitars there for the last four years. **How do you know, I asked, him, what to do with a guitar's electrics?** There are dozens of different models, each one different, and often wired to different colour standards.

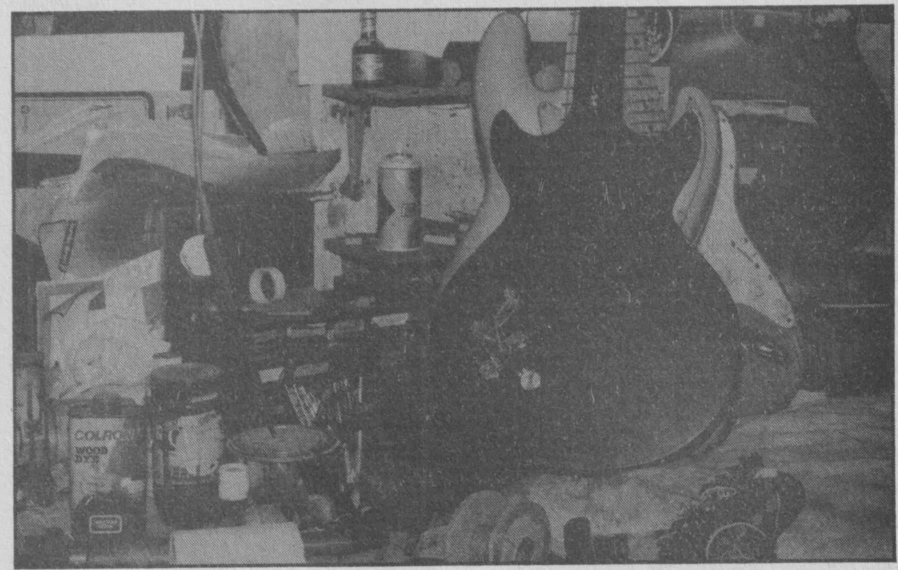
His first response was to produce a familiar-looking volume from behind the workbench. As it turned out, it was not the book I had at first thought, but a close relation: *Guitar Electronics: A Workbook*, published by the D.B. Music Co. of Ojai, California. This is by the same author, and is in some respects the same book, as *Guitar Electronics For Musicians*, available in the UK from Book Sales Ltd. It is older, and it is the bible of guitar electronics, and I am not going to enquire how it can be obtained, for reasons which will become clear later on.

"We refer to this book" said Mal. "This has the wiring of all the classic American guitars in it. And we also use the makers' own service manuals, which we get from the manufacturers."

What sort of work is he doing most of the time?

"Ninety per cent of the work we do here is on electric guitars" (although Andy's also have a reputation for building and repairing acoustic guitars). "It's mainly setting up the body. The body is the most important part, especially the neck. If the neck's no good, the guitar won't play properly. A cheap guitar can be improved quite a bit with expensive hardware, as long as the construction is sound and accurate in the first place. People like to modify guitars to suit their own preferences. There is really no set pattern to the type of alterations people want. It's a matter of personal taste. But you can't improve a really cheap guitar, especially an acoustic guitar."

What about customising upmarket guitars? I asked. You do hear of expensively modified guitars which actually lose their market value, because they are no longer in 'original' condition.



"People don't...well, the shouldn't... start customising a valuable guitar until they know they're going to keep it and play it seriously. Once a player has found a guitar he really likes (and it can take a few), he's not worried about what it will sell for. Guitars with a well-known name on the headstock do keep their value better than cheaper kinds, and if you do a good job of modifying a guitar which isn't actually a vintage, it's not going to put its value down."

And (this is a burning question) do people come to you with their favourite guitars and say, make me sound like Ritchie Blackmore? And what do you tell them?

"Yes, they do! It's alright if they know exactly what sound they're after, and roughly how to get it, but this includes knowing about effects and amplifiers as well as the guitar. If they don't know, I tell'em to go away and play some more guitars, go to a few music stores, get to know more about it. It's no good us just going ahead, because if we do something to their guitar, and it turns out to be not what they expected, they're very disappointed, and it's costing them money too."

"A bad guitar will hold you back, but you won't benefit from a really good one unless you are a better player."

What about the electrics? What goes wrong with them, and is it wise for the amateur to have a look?

"The commonest electrical problem is when someone has been poking about to see if something is wrong, and has made it worse! Sometimes there was nothing wrong with the guitar at all. A jack connector on the lead may have come unsoldered" (this is inevitable on cheap leads with right-angled jacks at one end), "or a volume pot was turned down, or a fuse had gone somewhere.

Guitar electrics, even simple ones, are very close packed, so it's very easy to melt a bit of insulation without noticing it, or overheat another component and burn it out."

"We're talking about people who don't understand electronic components here. If you do know one component from another, then it's safe to try repairing a fault. Use a small iron, 15W to 25W, and use the heat carefully. If you get a brittle joint, it could easily crack off if the guitar is banged or dropped.

"If it's complicated, or if you've got active tone controls, you don't want to tamper with it unless you have a circuit diagram. Guitars have mostly passive circuitry, driven by the amplifier. There

shouldn't be any danger of electric shocks, but if you are testing it with the amp connected, then make sure that the earth is properly connected on the amp, or it could electrocute you."

"Try to check with a wiring diagram before doing any tests, and check visually for shorts or loose connections. If it's excessive noise you're worried about, make sure that it isn't a cable, or scratchy pots which are causing the trouble. You can clean pots."

What about screening? I asked. I've seen some good quality custom built guitars which are not internally screened at all. (This is where electrically aware guitarists resort to tinfoil or silver-loaded paint in the body cavity).

"Many makers do good screening now. Fenders, of course...they have noisy single-coil pickups, with no screening at all. Gibsons are very good. Some guitars have a screening cover which you can see when you take the back off, others you can't see how it's screened. If the guitar is noisy, and you open it and you can only see the wires and components its not screened.

Now the bit about the aforementioned book. "Before we had that book" added Mal, "we just used to copy working guitars and make a note of where all the works are, and go by that. The book's not very accurate." I blanched. "But whenever we find something that doesn't match what we're working on, we just change it in the book, and then we know it's alright" he beamed. So much for a reliable reference source.

He fetched out a shapely Gordon Smith six string from a pile of valuable-looking instruments waiting for attention. "This one's by an English maker" he said "It's got beautiful electrics". He removed the cover on the back to reveal a neat, symmetrically organised nest of components. "There's a lot in there, but it isn't cramped up, and it's easy to get at."

Finally, I asked Mal, who is a freelance, how he got into the instrument repair business in the first place. Was it, like so many aspects of the music business, a matter of getting lucky and then learning on the job?

"No! I did a general instrument repair course, at Merton Technical College, two years! I got a grant. I did some evening classes first, but it never occurred to me that anyone would pay me to go and study it. It was great!"

"I started off by playing, originally, and then I tried altering my guitar. I was following a series in a music magazine, on how to improve your cheap guitar. That really got me started. I even built a guitar following a plan which was in another magazine." He extracted an ancient, yellowing fold-out plan from behind his workbench. "It's not very accurate. I mean, the frets aren't properly calibrated, or anything" — *(It wasn't published by Hobby Electronics, either — Ed.)* "but it gave me some ideas. A basic guitar is much easier to build than it looks."

"After I left college I worked part time. Then the job at Andy's came up, and I've been here about four years — Fiona's been here slightly longer. Grahame, another repairer, went to the London College of Furniture — you learn about all kinds of instruments, and then you can specialise. You can do electronics as well. It's a good thing to have a general knowledge of how instruments are made."

And how do you go about finding a good repair person?

"Find someone else who's used one! And see whether they are satisfied with the work. Try a reputable music shop, or somewhere they actually make guitars. Word of mouth is the main way of finding out. People won't always say, so & so's no good, but if you ask them whether they'd go back to the same person, and they aren't very keen, you can wonder!"

On the saw-dusty wall beside the spraying room is a display of signed photos from customers. Elvis Costello, Rockschool's Henry Thomas, Iron Maiden, Aswad, Status Quo — "we built the body of the bass before they did that Live Aid thing. It's a Travis Bean bass with a metal neck running through the body, and the body was cracking, so we built another one. One of the other repairers has got the old body!"

The other repairer concerned was Andy ("not *the* Andy"). Did they do the lead guitarist's famous green Telecaster? as well, I asked?

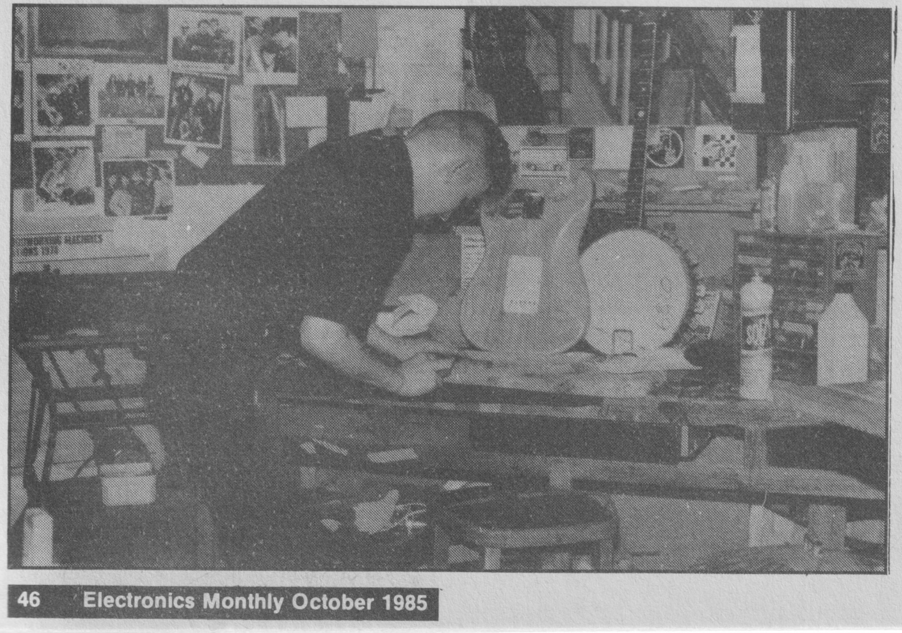
"Oh no! He did that himself on his kitchen table years ago! It was a sunburst, and he sanded it off and painted it green, and drilled that hole in the body" (some of the wood split away from the back) "and made changes to the bridge. Later he had it laquered when the green started falling off. I've got one just like it. They thought I was mad, here, cutting a Fender bridge in half! It's a great guitar. It works fine."

(He's mad.)

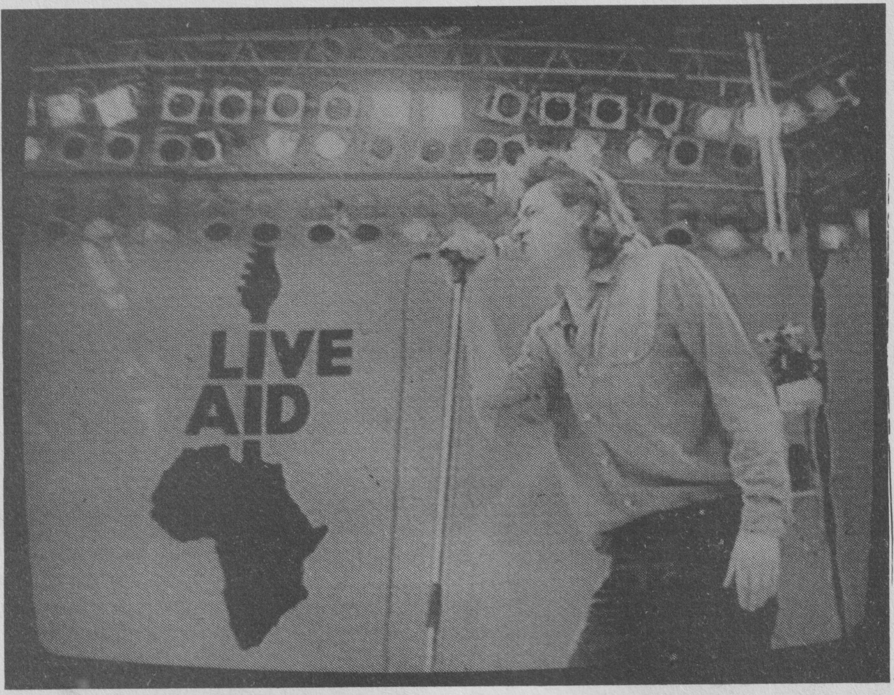
As Mal said, once you've decided to keep the guitar, you can do anything you like to get it the way you want it. But if you're not too sure — go and ask somebody like Mal. **Andy's Guitar Workshop** is at 27 Denmark Street, London WC2 (tel: 01 836 0899). They also have a branch (the original) in Newcastle.

And what about a final piece of advice for my colleague with the crooked neck?

"Buy a better guitar."



LIVE AID LINKS



Bob Geldof: the organiser who pulled the plug.

All screen shots by kind permission of BBC Television.

You've heard about the stars. Now, what about the workers?

A. Armstrong

LIKE LOTS of *Electronics Monthly* readers, I couldn't go to the Live Aid charity concert, so I did the next best thing and had it on all day and night. I settled down to enjoy the music, thinking no more about it until the picture went during The Who. 'Tut' I thought, but afterwards I wondered how they had coped with sending the programme to 1.5 billion people in 150 countries. So, here are some of the answers.

The rotating stage was in three sections (and for sound-wiring purposes, four, including the strip in front of the rotating bit). As a band finished playing, the stage rotated, putting a new section before the audience, while another section was being stripped and the third prepared backstage. The stage was powered by a huge motor, controlled by one man with a TV monitor, a pedal with a clockwise and anticlockwise mechanism and a speed control. The stage did its job perfectly.

But there were some problems: short notice, the number of acts, and the short changeover time between groups. "The programme was much bigger than when we started planning" said BBC Engineering Manager Mike Hill. "On Monday our Outside Broadcast unit was at Wimbledon, and the following Monday it was in Sandwich for the golf. We had 2½ days at Wembley to set up. We could have done with another day, and a bigger O. B. van for the linking. We had 2 miles of multicore cable for the stage sound — never mind the communications and the cameras."

The BBC were using two TV Outside Broadcast (OB) trucks, two Radio One stereo trucks and a mobile video vehicle with three VTRs for later playback. The power consumption was expected to be very high at one time they switched off the air conditioning in the trucks so that their circuit breakers would not trip out. This was with five technical staff and four production staff in one van. "The van came out barrel shaped" said Mike Hill.

A Link 8-camera OB truck as used by BBC TV at Live Aid.

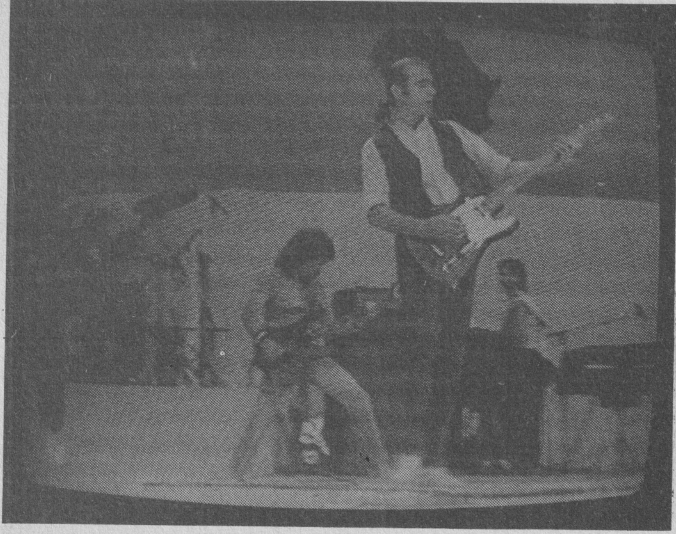
Pic: Link Electronics.

Mobile Control Room

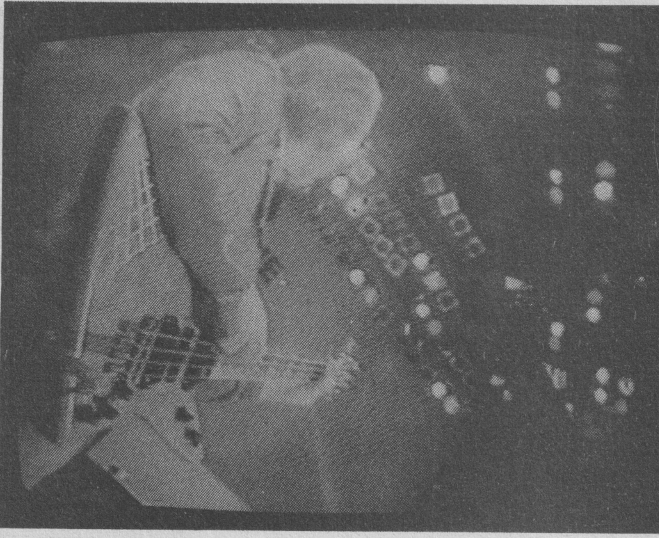
The bigger TV truck was a mobile control room with three sections: a camera control room, a sound desk, and an area for the production crew, with camera monitors. It normally runs four cameras, and can run another four. On this day, all eight were in use.

The real heart of the system was a smaller two-camera truck. It was from here that TV inserts from Australia, Japan and Eurovision, star interviews, and the USA For Africa concert were switched in and out of the world 'Simulcast'. The production was complicated, and, despite careful scheduling, no one could predict how long each changeover would take. "Sometimes we were killing time, and sometimes we were rushing around" said Mike Hill.

The large TV cameras in use were LDK5s, on wheeled pedestals, and the hand-held cameras were LDK14s. Two pedestal cameras were running across the front of the stage, one was on a 200 foot hoist just outside the rear of the stadium, one was set up on the 'mixing tower' in the crowd, one in the airship which observant people will have spotted circling the stadium, and one at the back of the stage on a "hothead"



Stars and stripes or, Shakin' All Over The World. The camera becomes a microphone.



The Who: sound and light in the Stadium, but a four minute black out for the rest of the world.

mounting. This was suspended from a lighting gantry (above the drummers!) and was remotely controlled by a cameraman behind the PA stacks on the stage. Not very comfortable!

But he was better off than cameramen using hand-helds on stage. As Concert Director Tom Corcoran said "The volume was so loud that the cameramen doing wide shots by the stacks couldn't hear anything in their headsets. The sound pressure level was so great that it was causing microphony in the cameras. The pick up tubes were vibrating and transmitting the sound, making coloured lines on the picture." (*And I thought it was the local CB – Ed.*).

Star interviews were taking place backstage and in the long gantry boxes high beneath the roof of the Stadium, some of them via portable Ikegami ENG (Electronic News Gathering) cameras. If the interviews were conducting sitting down, they were coming from the galley, and if standing up, from backstage, Tom said, and there were also some from the dressing room area.

Although the bands had strictly allocated times, the running order slopped about a bit. The broadcast had to meet two fixed points: the Wembley finale at 10pm (UK time) and the Philadelphia finale at 4am the following morning — and they made it. There were some victims. Howard Jones, for instance, only sang one number before the link up with Russia for the Russian group Autograph. But nobody knows exactly what happened next: either someone, somewhere (not the BBC, they stress) made a switching error, or else the Russians were not quite ready to show their band, because for the next minute or so we had a delightful programme about cherry-picking, but with the sound of the Russian pop group over it. No lip synch, either!

Then there were the power cuts. The longest one chose a prime time (during The Who, who had re-formed specially for this event) and lasted for nearly four minutes, but the first cut, just before David Bowie was not 'seen' on TV. It seemed to be an overload, but Mike Hill said "We were taking 350A, and we could have drawn 600A. The generator was a 150kW one, and we were drawing 80kW". A 'tired' fuse in the supply is blamed but the long break was caused by the generator (which replaced the mains) breaking down. Fortunately, the mains fuse was nearly repaired at the time. This incident gave rise to a curious effect.

When the Who came back on the air, sound and picture were out of sync for about fifteen seconds. Mike Hill explained "We were using the pictures we were sending to the USA until we got our own picture running again. They go across by satellite to America, and back again to us. You may have been getting our sound, and America's picture. The delay would be about .75 second." This was confirmed by Jeff Griffin of BBC Radio 1.

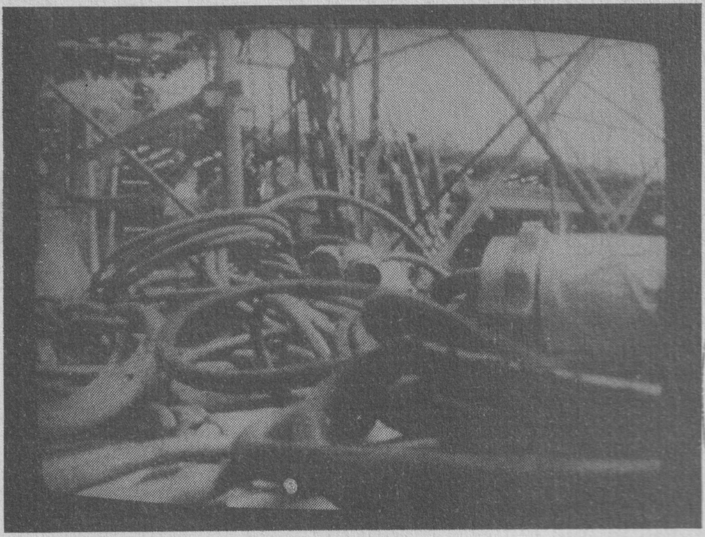
I timed the delay at about 1.4 seconds. I find these occasional occurrences interesting, but to Mike Hill, naturally, it is "acutely bad, if it went that wrong."

TV News

The BBC were not the only people filming: TV news coverage was co-ordinated by Visnews Special Services with two camera crews in the stadium. Using Sony BVP330 ENG cameras and 110 U-Matic video recorders, they filmed interviews, news and crowd shots. Bob Turner of Visnews said "We used about twenty tapes, which were either sent for editing, or were simply handed to customers — television stations and news agencies — on the spot! The U-Matic VCRs are not standard broadcast quality, but they are often used for news gathering — with the live atmosphere of news the difference between broadcast standard and non-broadcast U-Matics is not noticeable." You might recognise U-Matics as those whacking great videos you see in schools and elsewhere, which won't play ordinary video-cassettes. They are the standard type of professional VCR.

BBC Radio One took charge of all sound production, using two of their four stereo broadcast trucks (one from London and one from Glasgow)

Howard Jones gets a mere eight minutes to sit and do his stuff; the hand-held operator takes a break from chasing moving targets and moves in for an ear shot.



Heavy duty coils: a few yards of the estimated five miles of multicore cable in use on the site for sound, lights and cameras.

connected to two separate 'stage boxes' and the two mixers and PA rigs associated with them. The advantage of this was that the big seven-core multiway cables did not have to be swapped over as the stage rotated: this was the theory, at least. Each stage box was permanently connected to one outside broadcast van, and while one was in use, the other was being connected up to the microphones, DI feeds etc. for the next act. The two control trucks were able to 'leap frog', ie, while one was taking sound from the band on stage, the other was being set up for the next act.

The output from the stage boxes went to the sound desk in the 'scanner' (the mammoth BBC TV truck), along with sound feeds from the interviews in the gantry and backstage. Beside the radio/Tv sound, two other 'splits' were being taken from the stage boxes, one for the stadium PA and one for a smaller but important set of loudspeakers, the bands' stage monitors. Each stage box had 40 input lines.

Jeff Griffin, one of the producers, said "We had to turn around the bands within five minutes, which is unheard-off. Normally we can store information in the desk for total recall, and re-set the mixing desk in that way." But with no individual soundcheck on Friday, any re-setting had to be done on the spot by hand. 'Total recall' (usually) allows sound engineers to make settings at a soundcheck. Each microphone, instrument etc. can be correctly equalised for the venue, and the readings stored on disk. Later, readings can be displayed on a screen and reset without another soundcheck. Some settings can even be made automatically, although this is not used for live broadcasts.

The Radio people had problems when they lost some settings during the power cut — the first time the BBC had a mains and a generator failure at the same event. "We don't know what caused it" said Jeff Griffin "it could have been the oil filter. The man from the electricity board had just finished replacing the sub station fuse. We were off the air for five minutes. Five minutes is an eternity. It was out of our control."

The rotating stage may have given rise to another headache. The sound producers' master plan stated which of the 40 sound lines each microphone, DI Box, and instrument, was to use, for every band. But not all the right leads found their way to the right plugs, and precious time was spent trying to 'find' the correct lines. Then, when the stage was rotated for the next act, to quote Sound Producer Mike Robinson, "circuits appeared to have been disturbed." The large number of lines was needed so that, if necessary, several backing vocalists, a large drum kit (a drum kit can use fifteen mikes), all the instruments and two or three lead vocalists can be miked directly, as well as employing Direct Injection from some instruments, and still leave some emergency capacity: for instance, a spare microphone when a DI box on one of the bass guitars broke down.

A Minor Problem

Energetic organiser Bob Geldof caused a minor problem, pulling out his microphone cord as he leapt about. Unfortunately, the plug itself became dislocated. This in turn damaged the connector pins of one channel, putting it out of service for the rest of the show. The channel damaged was the lead vocal channel, so something else had to be dropped to make way for it. At another time, the sound from Paul McCartney's microphone failed to reach the BBC and the stadium PA — but came out of the stage monitors. Here, the audience helped out, by singing!

The sound mix for the stadium PA was different from the mix for the broadcast. Mike Robinson explained the reason for this. "The PA Mix is for the local acoustics. There could be little guitar in the PA mix, for instance, as a lot could be coming through the stage monitors, and a lot of vocal, to compensate for its quietness. The radio mix has to be suitable for radio loudspeakers. You lose the concert hall atmosphere, so we add a bit of reverb, and sound from the crowd. We use the same mikes, but split them with isolated parallels off the broadcast mobiles. Ninety nine times out of a hundred the radio mix will be different from the hall mix — just occasionally one mix is right for both."

The stadium PA was provided and run by Malcolm Hill Associates and their manufacturing company Hill Audio, using 120 M4 cabinets, two forty-into-eight channel J series 3F consoles at the mixing tower and two thirty-two-into-ten monitor Series 35 mixing consoles on stage (which use the low noise 5532 op amps which appear in EM now and then) all with a capacity of 150kW. They weren't able to call on this, though, as the local council (reacting to neighbourly complaints about the Bruce Springsteen concerts a week earlier) limited the sound pressure level to 100dB by the

And now from the USSR, Russia's own Autograph...or what? Viewers got the sound, but nobody knows who the cherry pickers in the pictures were.

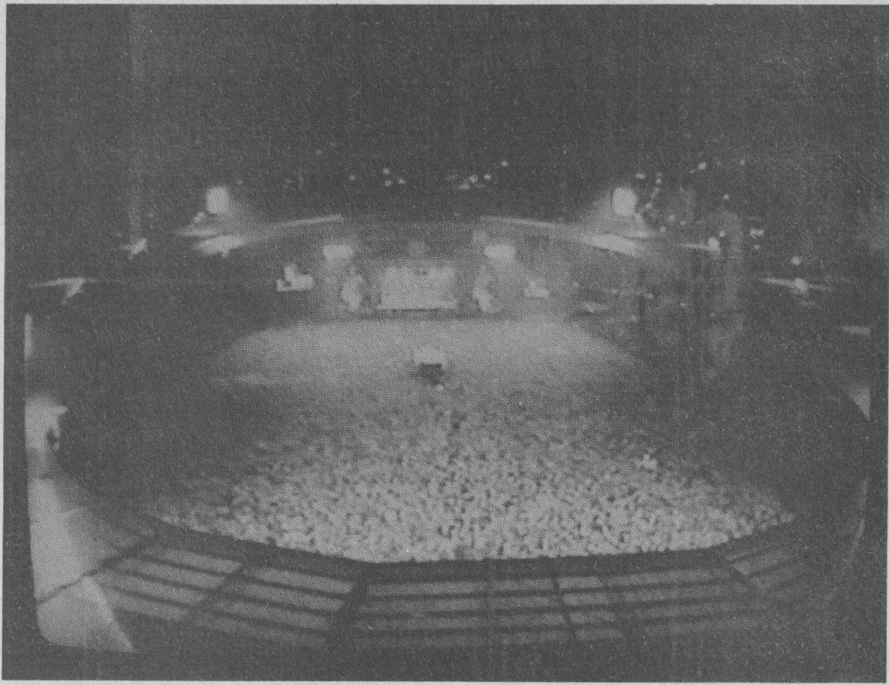
mixing desk and 93dB at the back of the stadium. The PA was consuming an average of 60-70kW — Malcolm Hill reckons that their usual average for a rock concert is 125kW.

The soundcheck the previous day had been for 108dB at the mixing desk and 102 at the back. "There is only about 4dB between acceptable levels of loudness for punters at a rock concert, and the levels outside" said Malcolm Hill "and people complain. The sound wasn't quite loud enough for real clarity everywhere in the stadium, although we've had nothing but praise for it: twenty two bands with no soundchecks."

The loudspeaker cabinets were suspended in an arc on either side of the stage from huge scaffolds. With no flooring, and each cabinet independent of the one beside it, the suspended speakers can be positioned so that each one couples in phase with the one next to it, instead of interfering with it. The scaffold and winch arrangement is easy to put and take down, although the equipment weighed about eighty tons.

As they were only informed about the sound pressure limit the evening before, another small stack of some ten cabinets was put up behind the mixing tower on a decay line from the main desk. They had no time for another sound check, so the delayed speakers were put in syn with the main stacks to the sound of the opening band, Status Quo! "We were still wiring it when they hit stage" said Malcolm Hill. "Somebody stood there and listened carefully to the snare drum, which has the sharpest sound, and then turned the knobs till the delayed speakers were in time with the front ones."

Sound from the main speakers travels at about 300 metres per second from the main speakers, and unless any speakers further back are fed with a signal which is delayed by an amount corresponding with the distance travelled by the sound from the main speakers, instead of reinforcing the sound a pre-echo will be added. Most delays are digital, using a 16 bit linear system.



Wembley Stadium at the finale of Live Aid UK, illuminated by the Stadium lighting. The control 'tower' with sound and light control desks is in the centre.

Reflected Sound

The oval stadium had presented a problem, he said. "You can't equalise for a big stadium like that without using several local systems. It would have been easier with a more powerful delay stack. There were areas where the overhang of the stadium comes round where the reflected sound would have been interfering with the direct sound, making it sound muddy. Right at the back it was fine, because it was only getting the direct sound."

With forty lines on stage, connecting everything between acts, listening on headsets over the sound of the stage monitors, was a rushed job. Sometimes artists came on before they were ready: Paul McCartney was one victim. "I thought the whole thing was remarkably fraught-less, considering that there were no rehearsals and that this had never been done before" said Robert Lingfield of Hill Audio. "My biggest problem was fighting the local taxi service off my walkie talkie — it was easier to get a taxi to Heathrow than talk to my own men."

Lighting is a vital part of any rock concert, but Live Aid was mostly in daylight, or twilight. Even so, the lights used included 350 Par 64 100W floodlights on stage gantries, out of sight at the top of the stage or concealed at the sides; a total of twenty 'follow spots' on stage and in the gantry around the stadium, and about forty Vari-Lites. These are pre-programmable computer controlled lights. The lights can provide any of one to ninety six colours, with beam patterns and movements. Each lamp is given a computer address via a thumbwheel on the lamp, and the colours and movements of that address stored in a central computer. The lamps are daisy chained with a cable; when an address goes down the line, the lamp with that address responds, giving instant changeover to any combination.

The lighting was being provided by Zenith Lights and co-ordinated by Peter Clark of Supermick — usually they are competitors. Two crews were provided, for stage and audience lighting, and it took five days to set up. Individual lighting plots were sent to lighting designers working for the groups, for approval — but any extravagant suggestions were put aside to keep it simple. This still required sixty lighting technicians, and thirty two follow spot operators, as well as special crew for the Vari-Lites, which were mounted on stage lighting trusses anchored to the roof. The floodlights were mounted six to a bar on ready-wired Socopex bars with 19-pin connectors. The main multicore cable went to a ninety-six way Avo (not the meter manufacturers) QM mixing board console at the control tower in the crowd.

The follow spot operators, seated high in the gantries around the stage and around the edge of the stadium, worked in shifts of 90-100 minutes each — apart from one man, who stayed in the gantry for twelve hours as a charity fundraising effort. Jeff Harold of Zenith explained "It is exhausting. You must concentrate on what the artist on stage is doing, all the time, and follow him closely; at the same time you have to listen to the lighting designer in your headset, telling you what colour to change to, and when to change. Each spot has six colour frames to choose from, in a magazine. As one slides out, the next colour slides in. And it was very hot — about 120 degrees up there."

"We had no breakdowns. We did lose a follow spot operator — he was moving a truck at the time! The lighting didn't begin to look really effective till 10pm." (Which, if you remember, was when the Wembley concert finished. I disagree, but I think that us telly-watchers saw more than those who were there, as the lights, mounted over twenty feet up in the roof of the stage, were only visible to those close to the stage.)

Returning to the beginning of this article, we have pictures and sound vanishing into five BBC mobile broadcast vans — but where did it go from there?

Satellites

Much of the news distribution was handled by Viznews who has a dedicated transponder service through Comsat and British Telecom International from Intelsat. This provided the service to the BBC, ITV and NBC in the States. From the BBC's OB trucks, the pictures were transmitted to the BBC Television Centre in Shepherd's Bush, then to the Telecom Tower in central London, and then to Viznews's establishment in Park Royal — not far from the Wembley Stadium! Vixnews the retransmits the signals via the Telecom Tower, via a satellite link to BTI's Goonhilly earth station in Cornwall, then by satellite to the Comsat earth station in Andover, USA, and via the Western Union Earth station and satellite to Philadelphia. In Philadelphia,, it was packaged for transmission on the Pacific Ocean Area satellite for Japan, Taiwan, S.E. Asia and Australia. Meanwhile, the same material was being transmitted by BTI's occasional TV service the Atlantic Ocean Area satellite for Europe, Africa and the Middle East, and from the Madeley Down earth station to the Indian Ocean satellite for Hong Kong, Western Asia and India.

While this was going on (earlier in the day) material was being received from Europe, Russia and Japan via satellite and European Broadcasting Union links, and (later in the day) material was coming in to London from Philadelphia for combining with the concert at Wembley before re-transmission as a worldwide broadcast.

Star on stage, operator in the scaffolding: will they do it again next year?

The three international television satellites are Intelsat Vs, the largest and most modern satellite in general use. The link from the JFK Stadium in Philadelphia to the earth Station in Los Angeles is a Westar, a domestic satellite serving the US only, and owned by Western Union. All the links carried stereo sound for BBC Radio One and ABC Radio in the States. The alleged satellite failures, said Viznews's Simon Patch, were due to power failures at the Stadia in Philadelphia and London.

So they didn't get away scot free.

Live Aid was seen by about one-and-a-half-billion people worldwide, and made £50 profit for the Ethiopean famine appeal, none of which would have happened if a large number of professional people had not donated a great amount of time and hardware in a coordinated effort the like of which has never been seen. Now quite a few local efforts are taking place — who knows, you may find yourself playing, or up a lighting gantry, even if it is not a 200 foot one. Or you might just want to make a donation to Band Aid, in which case, here is the address.

Band Aid Trust
c/o Stoy Haywood
8 Baker Street
London W1M 1DJ

SHORT CIRCUIT

Simple Temperature Alarm

Toh Eng Kiong

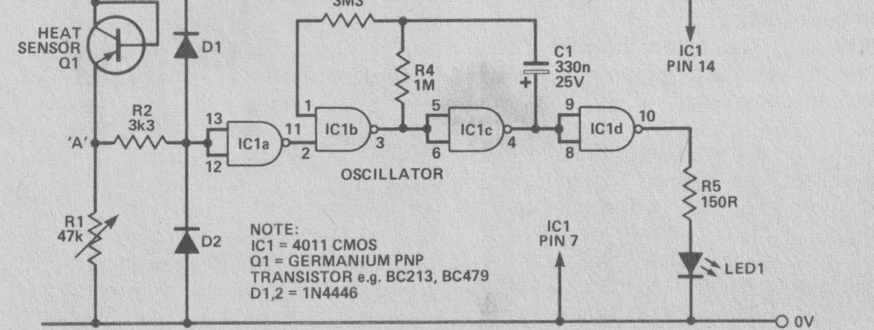
I have designed this circuit for use with my portable audio sound system (a tuner-tape and amplifier). The trouble with portable units is usually excessive heat generated by the power section after the audio system has been turned on for about thirty minutes.

The Temperature Alarm is made up of a single low power CMOS 4011 NAND gate. IC1a puts a high impedance at the forefront and switches the oscillator (from IC1b and IC1c and associated passive components) on or off in relation to the voltage at point a. The heat sensor is in fact wired around the germainium transistor Q1 and together with potentiometer RV1 form a potential divider. The sensing voltage can thus be adjusted within the sensing temperature range of the sensor. IC1d acts as a buffer driver to provide sufficient power to drive the LEDs.

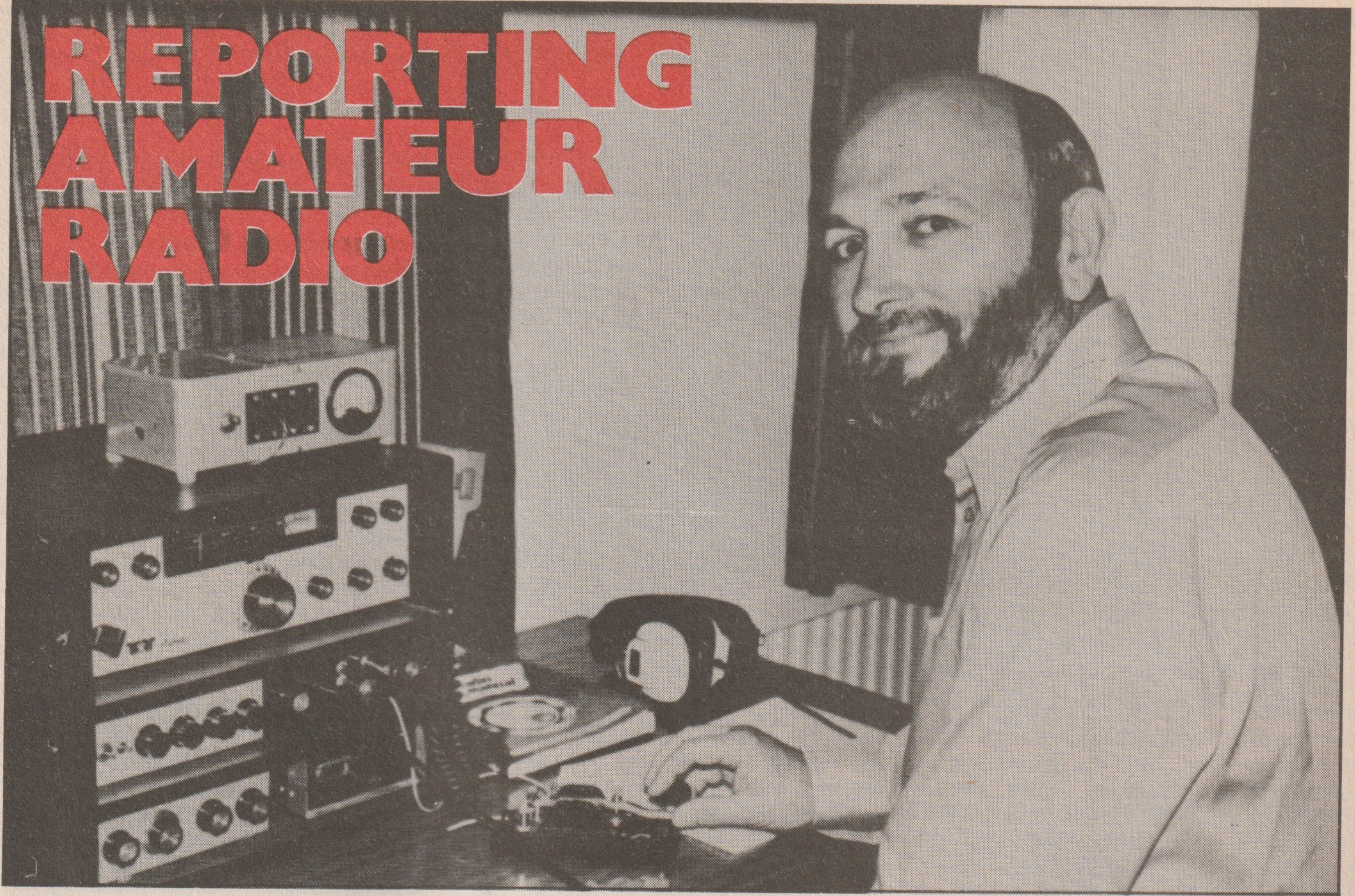
At switch-on, when the temperature is low, the LED blinks at around 2Hz or so, but once the alarm is activated, the LED glows continuously, indicating that the pre-set heat level has been reached. The heat sensor Q1 is soldered to a length of wire so that it could be placed some distance from the other parts of the circuit.

I tacked the heat sensor over the vent holes of the audio system where most of the heat is felt; this works very satisfactorily. I make it a point not to run the audio system too hot, by shutting it off when the alarm is activated — this might be after two hours of continuous operation.

I must add that I set the heat switching level of the alarm by intuition, not corresponding to any exact measurement.



REPORTING AMATEUR RADIO



Beginning a new column which will be talking about all aspects of amateur radio and reporting from around the scene.

Tony Smith G4FAI

WHEN MY FAMILY was young, I made my first simple, single-valve, shortwave radio from a kit. The literature promised I could 'listen to the world', but looking at the set, as constructed by my inexpert hands, I was rather doubtful about that!

To my amazement, I found I could hear powerful broadcast stations literally everywhere, in Australia, America, Russia, and so on. Then I discovered other, weaker, stations with people talking to each other over the air who, I learned, were known as 'radio amateurs'.

With a growing family, it wasn't possible to progress to more elaborate things at that time and the 'hoot-nanny', as the one-valver was called by my children, gave good service over a number of years.

The bug had bitten, however, and when my son was 13 I made a slightly more advanced receiver, this time with new-fangled transistors, specially to tune to one of the amateur radio bands. We sometimes listened together, and the day he said, "Can't we do that Dad?", I jumped at the opportunity, and within a short time we had both enrolled at our local evening class to study for the Radio Amateurs Examination! This course, usually over one academic year, gives a good basic introduction to electronics generally, apart from providing sufficient knowledge to successfully operate an amateur radio station.

For the record, we both passed the examination, and were given consecutive call-signs, my son receiving his shortly after his 14th birthday. We still didn't know a lot about the world of amateur radio and what we discovered then, and later, is now on offer through this column for the benefit of those following in our footsteps.

So Many Aspects

Who are radio amateurs? What do they do? These are questions to be answered in the months ahead. Various aspects of the hobby will be looked at in turn and, at the same time, news of current activities and developments will be reported, and their implications discussed.

Amateur radio is practised by people of all ages, male and female, from all walks of life, many of whom have no connection with radio and electronics in any professional sense. There are some of that ilk, but daily it is quite possible to find actors and doctors, gardeners and politicians, busdrivers, farmers, and students, all busy communicating with each other through the medium of the amateur radio wavebands.

It is, however, a scientific hobby, requiring a certain amount of successful study before a licence can be obtained, following which its practitioners have a choice of just talking to each other, constructing equipment, experimenting with antennas, or many other things. Some use the Morse code, some become keen on communicating with very low power transmitters. Some send their signals via amateur space satellites, and some bounce them off the moon.

Amateur television has its enthusiasts, as does RTTY — working with radio teleprinters. Signals can be aimed at, and reflected from, the aurora borealis, or from showers of meteors as they enter the earth's atmosphere. Some of these activities are so specialised they are virtually hobbies within a hobby, but newcomers to any branch are always welcomed, and advice and assistance is readily forthcoming.

Like most hobbies, amateur radio has its own jargon, customs, organised activities, and formal structure. For the un-initiated, listening to amateurs speaking on the air is sometimes like hearing a foreign language, but all will be explained. Whether you are a complete outsider just curious to know what its all about, a CBer thiking of expanding your horizons, someone already on the way to becoming an amateur, or even newly licenced but still a little 'wet behind the ears', there will be something to interest you here.

Passing Messages

At the turn of the century, amateur enthusiasts taking up the new 'wireless' telegraphy in the United States used it in a similar way to commercial users, ie, for passing messages on behalf of others. Networks were set up, and messages were often relayed from one station to another over hundreds of miles. The national amateur radio organisation of the USA, set up in 1914, reflected the extent of this activity by calling itself the American Radio Relay League (ARRL), a name still in use to this day.

Although still practiced in the US, such activities are not permitted in Britain and third-party traffic, as it is called, is only allowed on behalf of the Red Cross, St John's Ambulance, County Emergency Planning Officers, or the Police, in emergency situations, or during exercises conducted by them. A well organised national body, the Radio Amateur Emergency Network (RAYNET) trains for these eventualities, and has a well defined and special relationship with those who use its services.

A fairly recent change (1982) in the amateur radio licence conditions brought in a new concession for special event stations which are often set up with special call-signs at County shows, village fetes, exhibitions, and commemorative or other occasions. At such events, non-licenced persons, under the supervision of the station operator, may speak into the microphone to send simple greetings to other amateur stations.

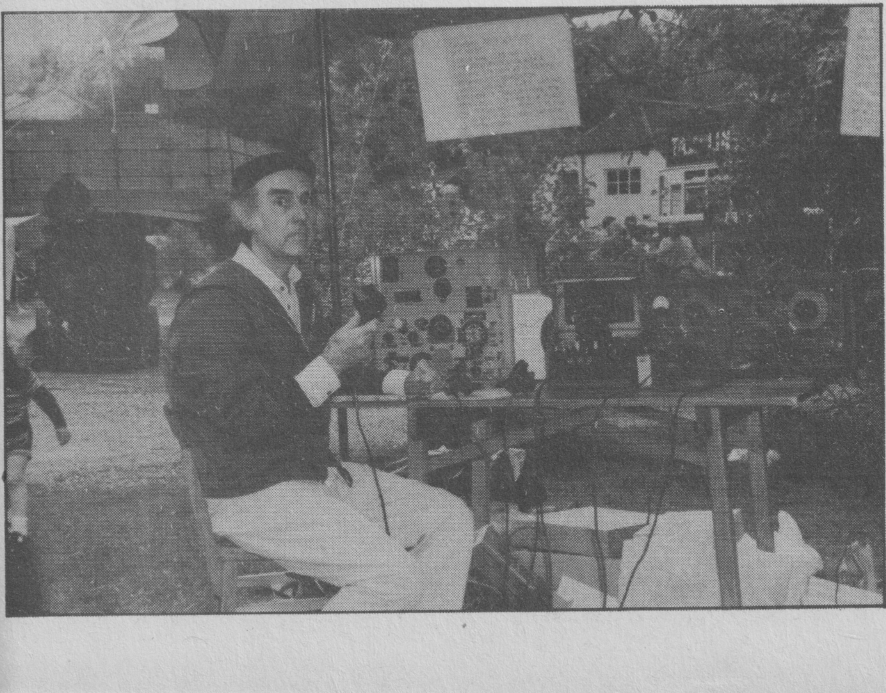
So far this has applied only to contacts with stations within the UK. Now, the Radio Society of Great Britain (RSGB), our own national society, has announced that following representations it has made to the Department of Trade and Industry, the official licencing body, the Department is seeking an agreement with the United States, Canada, Australia, New Zealand, and the Falkland Islands, to extend the 'greetings' concession to contacts with those countries.

The event which makes the most use of this facility is the annual Scout Jamboree-on-the-air, each October. Some countries already permit Scouts to send greetings to fellow members in other countries at this time, and the more the facility is extended the more meaningful the Jamboree will become.

Morse Code Testing

Although most amateurs communicate by speech nowadays, a large number still use the Morse code. Indeed, whether they intend to use the code or not, it is necessary for all who wish to work on the amateur bands, 10–160 metres, to pass the Morse test of twelve words per minute.

For many years this test was conducted by the Post Office. Nowadays it is the responsbility of British Telecom, but they are expected to relinquish testing in 1986. The DTI has asked both the RSGB and the City & Guilds Institute to submit proposals for taking on the work, and negotiations are continuing. It will be interesting to see the outcome. If the RSGB get the job there will inevitably be some who say they are not independent enough to test their own members.



Some years ago the City & Guilds changed the Radio Amateurs Examination to a multiple choice format, with a very mixed reaction from the amateur fraternity. If they take over from British Telecom, perhaps we will see a multiple choice Morse test as well!

Question Corner

Q. How much of a typical amateur radio station today is home-constructed?

A. Compared to earlier years, when it was difficult to obtain new equipment, not a lot. There is so much sophisticated commercially made equipment available today that, providing the money can be found, the temptation is to buy, not build. In fact, except for the highly talented, it would be difficult to make transmitting and receiving equipment to similar specifications.

Whether such sophistication is really necessary, is another matter. The basic requirements in putting out or receiving a satisfactory signal can easily be met by equipment which has been regularly in use for twenty years or even longer, and which has none of the modern features such as built-in memories, digital speech synthesis, and computer control.

Such equipment could be, and sometimes was, home-built, giving its owner great satisfaction and pleasure both in building and operating. Today, most home-building concentrates on station accessories such as antennas, antenna tuning units, SWR bridges, keyers, power supplies, and so on.

In one field, home-brewing does still produce transmitting and receiving equipment — in the field of low-power (QRP) operating. Simple transmitters putting out 2 watts, or even less, about enough to light a small torch bulb, have successfully communicated round the world, and some stations are completely home-made.

It is all a question of choice. Some still swear by valves, probably because they were brought up with them and understand them, and a soldering iron as their constant companion. Others are more than happy, owning and operating the latest 'black box' which has no user serviceable parts. There are plenty of other things to do in amateur radio if you don't want to make or repair your own equipment, so who is to judge what's right or wrong?

Questions please!

If you have any questions about amateur radio generally, why not write to me via the editor? I will answer as many as I can. I can, however, only answer through these pages, and space will not permit extended technical explanations except on rare occasions. ***(So no SAEs for this column, please — Ed.)*** I look forward to hearing from you.

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From next month ELECTRONICS MONTHLY is being merged with EVERYDAY ELECTRONICS. The new EE & EM will incorporate many projects by EM's regular contributors plus—more news, more component buying data, a brand-new beginner's series, more information on putting projects together and an efficient p.c.b. service, etc.

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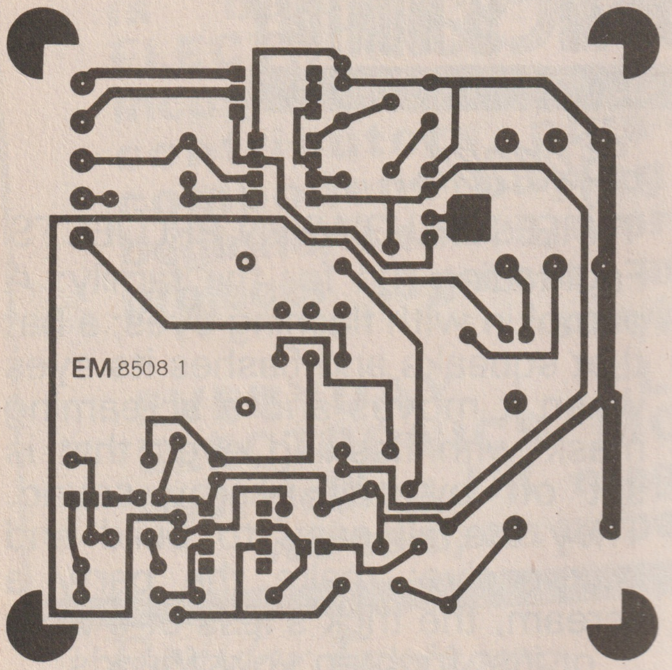
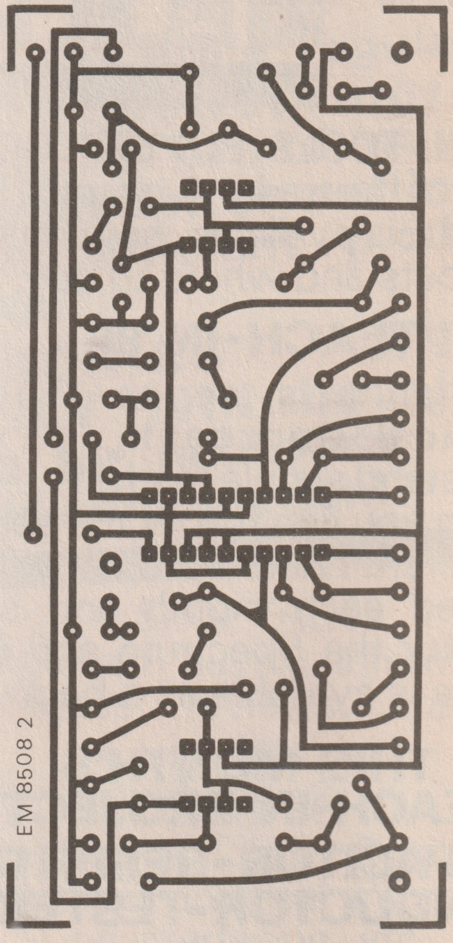
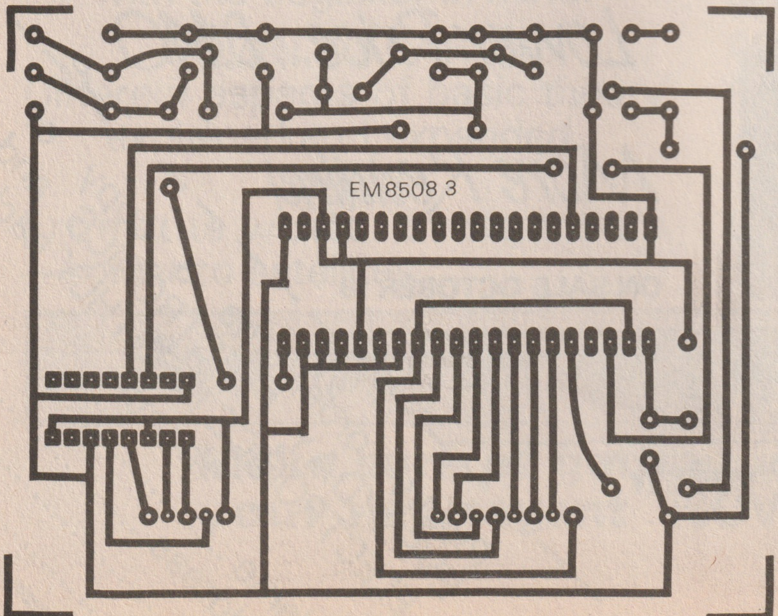


Above: the PCB foil for the Compressor Pedal. The arrangement of the switches is to fit the Maplin footpedal case used with the prototype.



Right: The PCB foil for the Computer Controlled Filter.

Below: The PCB foil for the Spectrum MIDI Interface.



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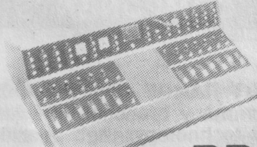

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Electronic MAILBOX

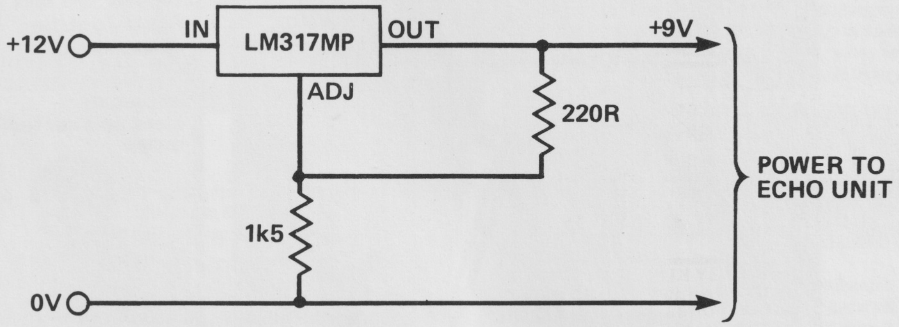
Vroomvroomvroomv..

I have just built the Echo Reverb unit from HE May 1982. This is designed to run off 240VAC, but I want to change it so that it will run off a car battery (12V). I have not had much experience with electronics and would be very grateful if you could send me details on how to convert it to run in my car.

Yours faithfully,
I Vandy,
Southampton.

In your car??? The mind boggles. The circuit should work, but probably at a reduced dynamic range. The circuit was intended to run at a higher voltage than can be obtained from a car battery. To make the modification, remove the mains transformer, rectifier, smoothing capacitor and voltage regulator. Substitute the circuit shown here, and connect the output to the holes in the board where the OV and output of the voltage regulator used to be connected.

The LM317MP is available from Cirkit, and the pin connections are shown in their catalogue. Tel. (0992) 444111. Finally, you have to readjust PR1, the bias preset.



Depth For Phaser

I have just ordered all the parts necessary for the Anti Phaser (EM February 1985) and noticed that it has no depth control, unlike most commercially-made units.

I was wondering if there was a simple modification that would give me this extra control — a simple mixing circuit/pot to balance the phased signal with the clean one or a control to vary the depth of the LFO waveform?

Certain songs benefit from different amounts of phasing and control over the depth would be a very good asset.

Any help you could give me would be very gratefully received.

Yours faithfully,
K. Winch,
Swansea.

There should be no difficulty in providing a depth control on this project. The frequency dependent negative feedback which introduces the notches is provided by R8 and R19. A linear dual gang potentiometer of about 470k in value is needed, with one gang connected in series with R8 and the other one added in series with R19. Maximum value would give minimum depth — minimum value would give maximum depth.

No Sound

Can you possibly tell me where I can get hold of the following semiconductors?

MN3011 multi tapped delay line.
MN3101 clock oscillator chip.
CLM6000 opto isolator with variable resistance output.

I've collected some smashing designs for musical effects units using these but having spent over six months looking for them I'm getting just a little fed up with forking out for catalogues. Being an impoverished student it seems a bit pointless spending a fortune on catalogues only to find I couldn't afford to build anything 'cos I'd already spent all my cash finding the bits! Therefore I'm grovelling for help.

May I also congratulate you on your marvellous mag. Keep the music projects flowing please.

Many thanks,
Dave Green,
Southbourne,
Bournemouth.

It looks as if these are specialised ICs intended for industrial use. Our investigations of the usual sources indicate that these ICs are not available in the UK at the moment, as the distributors are not handling them. This is a problem with specialised ICs. Tip: before you start building a project, check that you can obtain the main components, especially the ICs and transistors. If anyone can tell us more about these, or any other specialised music project ICs, we would like to hear from them.

Spectrum Interface

In November 1982 I received my first ZX Spectrum, and got an expert from our village to build me an interface from your feature in HE September 1982.

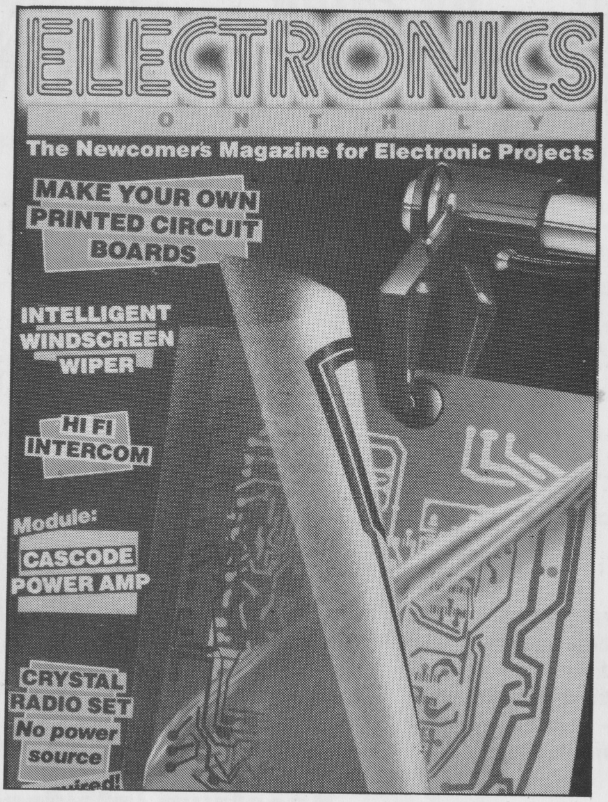
In August of 1984 I decided to rebuild the interface. When I completed it, I found that output line 5 was and is always set, and it cannot be switched off using the port 65503. My friend advised me to write and ask if this is a general fault and to find out if there is a way to counter this problem. I need the system to work my eight coloured lights and give a simple disco display.

Yours faithfully,
R.S.Dodds,
Spilsby,
Lincs.

We don't have any record of a general fault on this interface board. If your friend has an oscilloscope perhaps he could check whether the D5 input to the board is switching when the appropriate OUT statement is executed. To check more carefully, you could trigger the oscilloscope from the output of IC3 pin 6, as the only time that the state of D5 matters to the interface board is when the latch pulse occurs here.

If D5 does not switch then suspect the connector or the tracks in the Spectrum, or else suspect a solder bridge or a defective IC1. To check this, you could try exchanging the wire link to IC1 pin 4 with one of the other channels, and see if the fault moves with the connection.

BACK NUMBERS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